Session 2aAAa

Architectural Acoustics and Noise: Libraries, Media Centers, and Similar Spaces

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

Damian Doria, Cochair
Stages Consultants LLC, 75 Feather Ln., Guildford, CT 06437-4907

Chair's Introduction—8:00

Invited Papers

8:05

2aAAa1. Introduction and overview—Libraries as a building type and reflections of multipurpose trends that have an acoustical impact. Dennis Paoletti (Paoletti Consulting, 708 Foothill Dr., San Mateo, CA 94402, dpaoletti88@gmail.com)

Libraries have developed from vast storehouses of bookshelves and hard covered manuscripts and literary materials to vibrant community centers and true multipurpose facilities, acoustically. This paper will discuss the: history of libraries, politics of libraries, architectural design of libraries, acoustical design of libraries, and the future of libraries. Examples of library projects ranging from main urban public libraries and university library systems to a number of smaller, standalone local community libraries. will provide interesting insight into the variety of activities that go on today in libraries that challenge the best of our acoustical design sensibilities and controls. Libraries share the basic acoustical parameters of many other building types, e.g., open plan spaces, offices and conference rooms, cafes, and even digital/multimedia and performing arts facilities. Lessons learned from challenging consulting efforts, especially when budgets are limited, will be explored and discussed.

8:25

2aAAa2. Soundscape of the evolving library. Gary W. Siebein, Hyun Paek, Marylin Roa, Keely Siebein, Jennifer R. Miller, Gary Siebein, and Matthew Vetterick (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

Historically libraries were thought of as places of quiet reading and research. There were 2 general acoustical design approaches to reading rooms in libraries. One was to use large volume spaces with sound reflective materials that amplified every sound made in the room so that people would be self-conscious of the sounds that they made. A second approach was to design rooms with sound absorbent finishes that would reduce the spread of sounds from one part of the space to another. The evolving library of the future has many more complex functions and rich programs than the traditional library. Many of these encourage discussion, recreation, public involvement, experiencing multi-sensory media in addition to traditional books and a variety of other activities such as coffee shops, community meeting rooms, auditoriums, recording studios, listening rooms for amplified audio, interactive computer work stations, collaborative work/study areas where students can gather, talk, view and listen in active sessions all in close proximity to each other. Case studies of 2 libraries will be presented to identify links between soundscape design issues, architectural planning strategies and practical systems for designing the library of the future as it continues to rapidly evolve in concept.

8:45

2aAAa3. Recording room trends in libraries. Nicole Cuff (Acentech, 33 Moulton St., Cambridge, MA 02138, ncuff@acentech.com)

The use of libraries in communities and universities continue to evolve to where now there is a trend toward recording rooms in public and university libraries. One such space is a recording suite in a public library intended for the community to use, teens to seniors, with fully amplified bands next to study spaces, with control equipment available for patrons to use. Another in a private university is a podcast suite that overlooks a 4-story glass Atrium, which will be the new central meeting place on campus. The author even personally used recording studio space immediately adjacent to quiet study space at a private university library with their singing group to record an album. Some of the strategies used to make these spaces work together acoustically will be discussed.

9:05

2aAAa4. Acoustic conditions in two libraries at the University of Nebraska. William Spallino (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S 67th St. #210C, Omaha, NE 68182, wspallino@huskers.unl.edu) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Lincoln, NE)

University libraries serve many roles for students, from group gathering places to quiet study spaces; this makes analyzing and optimizing their acoustics very worthwhile. This paper reviews the acoustics of both main areas and small-group study rooms at the University of Nebraska—Omaha's Criss Library and the University of Nebraska—Lincoln's Love Library. Sound level meters were used to log sound levels in the two libraries over multiple days. Those data were then analyzed to understand the libraries' soundscapes through metrics like average sound levels, statistical sound levels, and occurrence rates of specific sound levels. Impulse responses and transmission loss of partitions were also measured. From those analyses, conclusions are drawn about the acoustic behavior of the libraries and its appropriateness for their purpose.

9:25

2aAAa5. West Hollywood Library—More than just another pretty space. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The West Hollywood Library is a 35 000 sqft branch of the County of Los Angeles Library, with construction funded by the City and operated by the County. It serves as a cultural and artistic community center, and includes a children's library, teen center, career development center, special collections, café, bookstore, two parking garages, community meeting room, and multipurpose public meeting room (for council and commission meetings, performances, and other uses). An ornamental wood ceiling, the wood stage in the multipurpose meeting room, the children's reading room, and the overall aesthetic presented interesting challenges for acoustics. Visual aesthetic and acoustic performance aspects will be discussed, along with references to various other library projects.

9:45

2aAAa6. Acoustics and AV technology of modern collegiate learning centers. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com) and William Roland (Roland, Woolworth & Assoc., Meridian, MS)

Many universities are creating new dedicated spaces for interactive learning that can host all subjects and a more modern and holistic approach to collegiate education. In order to achieve high performance facilities that mix classrooms, testing rooms, collaborative spaces, offices, and counseling all together, acoustical and audio visual requirements must be coordinated early in the project with the end user, with room to adjust to changing programming and technology. This paper presents 3 case studies of repurposed and new collegiate construction and outlines the types of program requirements and the approach to acoustics and audio visual for these projects.

10:05–10:20 Break

10:20

2aAAa7. Acoustical design for libraries in schools to comply with ANSI S12.60. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com)

In K-12 schools, the library functional space is now usually called the media center, as this space houses and disseminates information that is stored on both print and digital media. These spaces have fewer sound absorbent book stacks than were found in previous print-only libraries. Instead, computer workstations for student access to local and web-based material are provided. Often these facilities are used as group learning spaces, with several group classes occurring simultaneously. These changes to the traditional school library space present challenges to the acoustical design. Another challenge is that in a number of school districts across the US, a media center must be designed to meet the requirements of ANSI/ASA 12.60-2010/Part 1, "American National Standard Acoustical Performance Criteria, Design Requirements and Guidelines for Schools, Part 1: Permanent Schools," also known as the "Classroom acoustics standard." This is because a media center can be defined as a core learning space that is subject to that standard, with its requirements for low noise and reverberation, and for higher sound isolation from adjacent spaces. Several case studies are presented which highlight the design challenges for modern school media centers, and the successful design solutions.

10:40

2aAAa8. Considerations for media center design in unconventional environments. Melvin Saunders (None, 1601 Elm St., Fl. 33, Dallas, TX 75201, melvin.saunders@saundersassoc.com)

The consolidation and simplification of media in production environments has rapidly changed the way that many businesses develop content. Rather than outsourcing media development, many clients have undertaken the task to develop content for both internal and external consumption by building in-house media centers. This paper will focus on some of the lessons learned for these types of projects from both a remediation and new construction process. Acoustical solutions for projects of varying budget and scope including room acoustics, sound isolation, and HVAC noise abatement will also be explored.

11:00

2aAAa9. A church space that could be a library, media room, meeting room, performance space, or dining room—or with proper room acoustic modifications—An effective sound reinforcement system including audio recording, and suitable video recording and projection it could function as all of the above noted spaces. Robert C. Coffeen (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, bob@rccoffeen.net)

In a nearby church that was recently visited a space was viewed that is nicely appointed from an aesthetic viewpoint but primarily due to room acoustics, noise issues, and lack of proper sound and video systems, it did not seem to adequately serve any of the functions noted by the title of this presentation. The church is considering a renovation of this space. This presentation will describe how the space can serve all of the noted functions with relatively simple and easy to operate acoustic, sound system, and video modifications without significantly disturbing its current and desirable appearance.

11:20

2aAAa10. Rethinking libraries—Three case studies. Evelyn Way (Stages Consultants LLC, Chicago, IL) and Damian Doria (Stages Consultants LLC, 75 Feather Ln., Guildford, CT 06437-4907, damianjdoria@gmail.com)

As the nature of library usage changes, traditional book stacks and individual workspaces are giving way to group activities, informal meetings, classes, and makerspaces. In the past, distraction was managed not through the acoustic environment, but operationally through behavior enforcement and having separate physical spaces for different activities. Now excessive reverberation and inappropriate background noise levels significantly inhibit the usability of a library. Conversations from the café must not disturb small group work areas or instruction on the three-dimensional printer, community meetings need to happen at the same time as classes on digital photo processing, and childrens' story time occurs while the local entrepreneur researches how to register their business. We examine the acoustic design for a community library as collaboration center, a community library as recreation center, and a school library integrated into the common space.

11:40

2aAAa11. Acoustic design of today's library. John C. Swallow and Michael J. Wesolowsky (Swallow Acoust. Consultants Ltd./ThorntonTomasetti, 23 - 366 Revus Ave., Mississauga, ON L5G 4S5, Canada, jswallow@thorntontomasetti.com)

Modern libraries are very different from past libraries, offering multiple services and having wildly different acoustic requirements. Reviewed are new uses and functional requirements: more than just a traditional information resource, the library features information sharing, collaboration, dissemination and production as a media centre. A brief history of library design is provided. Over decades, library design philosophy has evolved which has provided its own challenges. Modern building design presents more challenges; adding in the new users and uses—which even vary by age group, the acoustic designer now faces enormous challenges. Acoustic design measures are reviewed and found wanting: new measures are needed. Case studies show the variety of uses, novel design solutions and highlight how a detailed knowledge of actual users and uses is required to allow all the functions to more-or-less comfortably co-exist.

TUESDAY MORNING, 14 MAY 2019 EXHIBIT HALL, 8:00 A.M. TO 12:00 NOON

Session 2aAAb

Architectural Acoustics: Student Design Competition (Poster Session)

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

Andrew N. Miller, Cochair
Bai, LLC, 4006 Speedway, Austin, TX 78758

The Technical Committee on Architectural Acoustics of the Acoustical Society of America with support from the Robert Newman Student Award Fund, the Wenger Foundation, and the National Council of Acoustical Consultants is sponsoring the 2019 Student Design Competition that will be professionally judged at this meeting.

The competition involves the design of a new municipal building including a court room and a community hall.

The submitted designs will be judged by a panel of professional architects and acoustical consultants. An award of USD$1600 will be made to the submitter(s) of the design judged "first honors." Four awards of US$800 each will be made to the submitters of four entries judged "commendation."

Session 2aAB

Animal Bioacoustics and Signal Processing in Acoustics: Bioinspiration and Biomimetics in Acoustics I

Rolf Müller, Chair

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

Chair's Introduction—8:30

Invited Papers

8:35

2aAB1. Most animals hear acoustic flow instead of pressure; we should too. Ronald Miles (SUNY Binghamton, Dept. of Mech. Eng., Vestal, NY 13850, miles@binghamton.edu)

The majority of animals that hear sound do so by detecting the minute fluctuations in the velocity of the medium. They do this by sensing the deflection of thin hairs that are driven by viscous forces in the acoustic medium. This is in contrast to the detection of sound pressure, as is used in some animals, including humans. Nearly all microphones respond to sound through the use of a thin diaphragm or membrane designed to experience a net fluctuating pressure in a sound field. Here, we explore technologies for achieving precise detection of sound using a mechanical structure that is driven by viscous forces associated with the fluctuating velocity of the medium. In one example, this has been shown to result in a directional microphone with flat response from 1 Hz to 50 kHz [Zhou *et al*., "Sensing fluctuating airflow with spider silk," *Proc. Natl. Acad. Sci*., 201710559(2017)]. Methods of creating extremely compliant, viscous-driven velocity sensors are discussed along with a new technique for capacitive transduction to achieve an electronic output.

8:55

2aAB2. Animals can measure acoustic flow; with the help of MEMS, we can too. Stephane Leahy and Ahmed Abdelaziz (Soundskrit, 780 Ave. Brewster, Ste. RC-016, Montreal, QC H4C 2K1, Canada, stephane.leahy@soundskrit.ca)

Animals measure acoustic flow by using thin hairs. These hairs are essentially mechanical structures that are driven by viscous forces. Manufacturing thin hair-like structures is challenging, but we can think of other mechanical structures driven by viscous forces that can be manufactured more easily. Here, we discuss how MEMS technology can be used to fabricate a thin porous membrane and electrodes for measuring acoustic flow. We discuss the design trade-offs between performance, manufacturability, and product integration. For example, from a performance perspective, it is desirable that the membrane be as thin as possible, but from a manufacturing perspective, it is easier to make it thicker. We present preliminary results on early prototypes and discuss avenues to create a MEMS acoustic flow sensor that could be found in your smartphone within the next few years.

Contributed Papers

9:15

2aAB3. Emission baffle deformations in bat biosonar and biomimetic systems. Liujun Zhang (ME, ICTAS II, Virginia Tech, 1075 Life Sci. Circle, Blacksburg, VA 24060, sdujune@gmail.com), Luhui Yang (Shandong Univ., Jinan, China), and Rolf Müller (Virginia Tech, Blacksburg, VA)

Old-World leaf-nosed bats (Hipposideridae) and horseshoe bats (Rhinolophidae) are two families of echolocating bats that emit their biosonar pulses through nostrils which are surrounded by elaborated baffle shapes ("noseleaves"). Prior work has shown that the noseleaves change shape in synchrony with ultrasound emission. These deformations involve at least two noseleaf parts with static (geometric) as well as dynamic complexity. To investigate how these deformations could be used in bioinspired systems, data has been collected from bats recorded with microphone/camera arrays as well as a biomimetic sonar head with a biomimetic noseleaf model inspired by Pratt's roundleaf bats (*Hipposideros pratti*). Both sets of experiments have produced qualitatively similar results that show an emitted wave field which depends on frequency, direction, and also time. Furthermore, the results obtained with the biomimetic noseleaf have demonstrated that noseleaf deformations during ultrasound emission can result in an enhanced information-encoding capacity. Current work is aimed at establishing whether this enhanced coding capacity can be utilized to improve performance, especially in sensing tasks that are related to the natural environments in which the bats' biosonar systems operate. It also needs to be established how much detail needs to be mimicked in an engineered system to harness these effects.

9:30

2aAB4. Hipposideros pinna motion patterns as an inspiration for dynamic biomimetic sensing. Pennine Jiu, Anorexia Yin, and Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Circle, Blacksburg, VA 24061, peiwenq@vt.edu)

Many bat species, e.g., among the horseshoe bats (Rhinolophidae) and Old-World roundleaf-nosed bats (Hipposideridae), have highly mobile pinnae. Experiments with biomimetic reproductions of the pinnae shapes/motion patterns in these species have demonstrated a dynamic enhancement of the performance in acoustic direction-finding paradigms. It could hence be hypothesized that reproducing the full range of pinna mobility patterns in bats will allow realizing the full range of the animals' biosonar capabilities. Pinna motions in rhinolophid and hipposiderid bats have been shown to fall into two distinct categories, being either rigid rotations (i.e., changes in orientation but not in the shape itself) or deformations that change both pinna orientation and shape. To characterize the variability in the rigid rotations, landmarks on the pinnae of hipposiderid bats have been tracked using stereovision. From this kinematics data, an axis-angle representation of the pinna rotations was estimated. The results showed that the axes of pinna rotations can scatter widely in orientation, covering a range of directions that extends 180 degrees in azimuth and 180 degrees in elevation, about 40 times larger than the estimated error of the employed method. Hence, biomimetic reproductions of bat ear mobility should explore how to make use of such a variability.

9:45

2aAB5. Reducing bats' pinnae deformation complexity for a biomimetic reception baffle dynamics. Jia Guo, Andrew Kurdila, and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., 144 Durham Hall, 1145 Perry St., Blacksburg, VA 24060, jguo18@vt.edu)

Horseshoe bats are known to have about 20 muscles on each pinna that allow the animals to deform their pinnae during echo reception. Simplified biomimetic reproductions of this reception baffle dynamics have demonstrated the encoding of additional, useful sensory information. However, reproducing the full complexity in the actuation of the bat pinna shapes by virtue of numerous muscles poses a daunting challenge. Furthermore, it remains unknown which level of complexity is necessary to realize the functional advantages of the bats' pinna dynamics. To address this issue, the current work has focused on non-rigid deformations of the pinnae. 3D kinematic data for a dense grid of landmark points distributed over the pinna surface has been used as input for manifold learning algorithms designed to reduce the dimensionality of the motion. The results of this analysis suggest that the deformations of the pinna can be described in spaces that with much lower dimensionality than the original kinematic data. In addition, a kernel method has been implemented to generate a reduced model of the pinna deformations. Ongoing work is aimed at understanding how these low-dimensional descriptions of the pinna deformation relate to muscular coordination patterns in the pinna and its dynamic functional characteristics.

10:00–10:15 Break

10:15

2aAB6. Fast moving biomimetic ears. Xiaoyan Yin and Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir., Blacksburg, VA 24061, xiaoyan6@vt.edu)

Like many other mammals, bats are known to rely on spectral signatures to determine the directions of incoming sounds, especially in elevation. Because of its simple hardware and computational realizations, this principle could also be suitable for small, parsimonious biomimetic sonar systems. However, the biosonar systems of bat species in families such as the rhinolophids and hipposiderids often—but not always—concentrate most of the emitted energy within a narrow frequency band in order to pick up prey-induced Doppler signatures. It remains unclear how the use of such narrowband biosonar pulses could be reconciled with direction-finding based on spectral signatures. A possible solution to this paradox could be provided by fast pinna motions in these animals that have been shown to produce readily perceivable Doppler-shift signatures in biomimetic reproductions. In the current work, such a biomimetic pinna with fast deformations has been used to map Doppler-shift signatures as a function of direction. For this purpose, the Doppler-shift signatures were clustered based on the similarity of their spectrogram-representations. Mapping the signatures' different cluster associations into direction space resulted in contiguous patches. Hence, it should be possible to obtain stable estimates of target direction based on the received Doppler signatures. <audio controls = "controls" style = "display: none;" > </audio> <audio controls = "controls" style = "display: none; " > </audio>

10:30

2aAB7. Coordination of sonar emitter and receiver dynamics inspired by bats. Shuxin Zhang (Virginia Tech Int. Lab.,School of Phys., Shandong Univ., Shanda South Rd. No. 27, Jinan, Shandong 250100, China, shuxinsduvt@yahoo.com), Yanming Liu (Key Lab. of High Efficiency and Clean Mech. Manufacture, School of Mech. Eng., Shandong Univ., Jinan, Shandong, China), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Recent biomimetic sonar systems have started to mimic the mobility seen in the emission and reception baffles of bats (i.e., "noseleaves" and pinnae) in greater detail. Some of these efforts have included rigid rotations as well as non-rigid shape changes. Since some bat species are known to actuate their noseleaves during pulse emission and their two pinnae during pulse reception, it may be hypothesized that coordination between the dynamic behaviors of these baffle structures is important to system-level biosonar performance. However, it remains to be determined how the complicated rigid and non-rigid changes to the noseleaves and pinnae of bats are coordinated in the animals' biosonar behaviors. To shed some light on this dynamic system integration in bat biosonar, we have conducted experiments with a species of hipposiderid bat (Pratt's roundleaf bat, *Hipposideros pratti*) where at least 24 landmark points have been placed on the noseleaf and both pinnae in order to track the non-rigid shape changes in all these three structures simultaneously. A canonical correlation analysis has confirmed that the motions of the noseleaf and both pinnae are highly correlated with each other. Work to identify the nature of these relationships using techniques from nonlinear dynamics is currently underway.

10:45

2aAB8. Pulse-train time structure for dynamic biomimetic sonar. Yanming Liu (Key Lab. of High Efficiency and Clean Mech. Manufacture, School of Shandong Univ., Shan Dong Univ., 27 Shanda Nanlu, Jinan 250100, Shandong 250100, China, larry.young.ming@gmail.com), Shuxin Zhang (Virginia Tech Int. Lab., School of Phys., Shandong Univ., Jinan, Shandong, China), and Rolf Müller (School of Mech. Eng., Virginia Tech, Blacksburg, VA)

The biosonar pulse trains of echolocating bats can have pronounced time patterns. A well-known example are adaptive increases in pulse repetition rates seen when certain bat species approach their prey. Furthermore, the production of pulse groups has been reported to create pulse-timing diversity outside of prey pursuit in several bat species. In order to replicate the biosonar capabilities of bats, a biomimetic sonar system may have to integrate this level of adaptivity. If the system also mimics the peripheral dynamics, i.e., shape deformations of the noseleaves and pinnae, in species such as the horseshoe bats, the question arises how the pulse timings and the shape deformations should be best coordinated. To investigate this issue, synchronized recordings of noseleaf motion and pulse emission patterns have been obtained from hipposiderid bats. The noseleaf deformations have been characterized (clustered) based on reconstructed trajectories of five landmark points. Similarly, the corresponding biosonar pulse trains were classified based on similarity metrics developed for neural spike trains and are based on cost assigned to that operations that are needed to be performed to transform one spike pattern into another. The relationships between distances in the noseleaf motion and pulse-train domains are the subject of ongoing research.

11:00

2aAB9. Biosonar inspiration for radar waveform design. Gregory E. Coxson (Elec. and Comput. Eng., U.S. Naval Acad., M/S 14B, 105 Maryland Ave., Annapolis, MD 21402, gcoxson@ieee.org)

Bat biosonar offers a natural source for biomimetic design of radar waveforms with inspirations falling into two categories: (1) biosonar principles similar to ones already employed in radar and (2) principles used by bats that operate in ways not yet understood or not yet embraced yet for radar. This second type offers the possibility for driving radar innovation. Hyperbolic frequency modulation (HFM) waveforms were one of the first aspects of bat biosonar to catch the attention of the radar community. Relative to the popular linear frequency modulation (LFM) waveform, HFM is less vulnerable to Doppler shifts from targets' relative velocities. Another promising possibility for biomimetics is the remarkable ability of some bats to resolve closely-spaced objects using acoustic frequencies on a small platform. Bats' ability to operate in swarms is of interest to radar designers concerned about mutual interference between ships or planes using the same waveforms and frequencies. As bat researchers approach an understanding of the mechanisms behind these, and other, abilities of bats, their results will find a ready audience. This talk will discuss aspects of bat biosonar offering payoffs for radar design, including but not limited to improved resolution, Doppler tolerance and mitigation of inter-system interference.

11:20

2aAB10. Cochlea-inspired sonar signal processing. Bryan D. Todd (Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave., Panama City Beach, FL 32407, Bryan.D.Todd@navy.mil) and Rolf Müller (Virginia Tech, Blacksburg, VA)

Many attempts have been made to model the signal transformations that occur in the mammalian cochlea, in particular with respect to modeling human hearing. Cochlea models that have been developed in the last few decades fall into several categories, e.g., electrical analogs, digital filter models, and incorporate different levels of complexity from entirely linear to strongly nonlinear. Similarly, the implementations used have included various numerical approaches as well as analog and digital hardware architectures. Mammalian hearing systems are known to be highly capable when it comes to analyzing sound from complex natural environments and soundscapes. It may hence be hypothesized that the signal transformations that occur in the cochlea play a critical role in facilitating these capabilities since the cochlea is positioned at the critical first stage of auditory signal encoding. The goal of the research presented here is to evaluate the signal transformations of the mammalian cochlea in the context of sonar signal processing, especially for automatic target recognition in difficult environments such as reverberant shallow water. The cochlea of bats could proof a suitable model for these applications since many bat species are known to be capable of solving demanding sonar tasks under difficult conditions and cluttered environments.

Session 2aBA

Biomedical Acoustics and Signal Processing in Acoustics: Cardiovascular Ultrasound: Imaging and Therapy I

Kevin J. Haworth, Cochair
University of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209

Jonathan A. Kopechek, Cochair
University of Louisville, 2301 S Third St., Lutz Hall, Room 400, Louisville, KY 40292

Invited Paper

8:30

2aBA1. Left ventricle blood flow patterns in a mouse model of temperature sensitive sodium channelopathy. Jeffrey A. Ketterling (Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., New York, NY 10006, jkettermeister@gmail.com), Orlando Aristizabal (Skirball Inst. of Biomolecular Medicine, NYU School of Medicine, New York, NY), Akshay Shekhar (Leon H. Charney Div. of Cardiology, New York Univ. Langone Health, New York, NY), Billy Y. Yiu (Dept. of Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada), Colin K. Phoon (Div. of Pediatric Cardiology, Hassenfeld Children's Hospital at NYU Langone, New York, NY), Glenn I. Fishman (Leon H. Charney Div. of Cardiology, New York Univ. Langone Health, New York, NY), Alfred C. Yu (Dept. of Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada), and Ronald H. Silverman (Dept. of Ophthalmology, Columbia Univ. Medical Ctr., New York, NY)

Sophisticated methods of visualizing and analyzing the complex blood flow patterns in humans have been available for a number of years on ultrasound machines. To date, these methods have not been applied to healthy mouse models or mice with cardiac defects. Here, a high-speed plane-wave imaging approach was used to study left ventricle blood flow patterns in a model of sodium channelopathy using mice lacking fibroblast growth factor homologous factor 2 ($Fhf2^{KO}$) and littermate wild-type (WT) controls. FHF2 binds to the cytoplasmic tails of voltage-gated sodium channels in mouse cardiomyocytes and modulates channel inactivation and cellular excitability. $Fhf2^{KO}$ mice have impaired cardiac sodium channel function that when body temperature rises causes a severe reduction in cardiac sodium currents, cardiomyocyte excitation, and conduction failure. The effect is reversible as body temperature returns to normothermia. $Fhf2^{WT}$ and $Fhf2^{KO}$ mice were imaged with a Verasonics Vantage 128 using an 18-MHz linear array with a 30 kHz absolute plane-wave transmission rate. The mice were supine on an imaging platform and a warm-air source was used to raise the body temperature. Surface ECGs were continuously recorded throughout the duration of the experiment. Data were post-processed using a least-squares, multi-angle Doppler analysis approach to obtain vector-flow estimates at all pixel locations. Vortex patterns, flow rates and ECG signals were compared between the $Fhf2^{WT}$ and $Fhf2^{KO}$ mice.

Contributed Papers

8:50

2aBA2. Live color encoded speckle imaging platform for real-time complex flow visualization *in vivo*. Billy Y. Yiu (Elec. and Comput. Eng., Univ. of Waterloo, 250 Laurelwood Dr., Waterloo, ON N2J0E2, Canada, billy.yiu@uwaterloo.ca), Mateusz Walczak, Marcin Lewandowski (Inst. of Fundamental Technol. Res., Warsaw, Poland), and Alfred C. Yu (Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

Complex flow patterns carry valuable diagnostic information but they cannot be visualized in a time-resolved and intuitive manner in real-time. In this work, we present the first real-time scanning platform for a high frame rate ultrasound technique called color encoded speckle imaging (CESI) and its use in visualizing blood flow dynamics *in vivo*. CESI visualizes these dynamics through duplex rendering of flow speckle motion and color-coded Doppler velocity estimates. Its live implementation was achieved by integrating a 192-channel programmable ultrasound front-end module, a 5 GB/s capacity data streaming link, and a series of computing kernels implemented on the graphical processing unit (GPU) for beamforming, Doppler processing and display rendering. A slow-motion replay mode was also included to offer coherent visualization of CESI frames acquired at sub-millisecond resolution (0.3 ms). The live CESI scanning platform was found to be effective in facilitating real-time image guidance (>20 fps). *In vivo* pilot trials also showed that live CESI, when operating in replay mode, can temporally resolve the formation and dissipation of recirculation at the carotid bifurcation and can reveal flow dynamics in the brachial vein during a fist-clenching maneuver. Overall, live CESI has potential for use in routine *in vivo* investigations that seek to identify complex flow dynamics in real-time and relate these dynamics to vascular pathophysiology.

9:05

2aBA3. Design of carotid bifurcation phantoms for integrative imaging investigations of arterial wall and flow dynamics. Adrian J. Chee, Billy Y. Yiu, and Alfred C. Yu (Univ. of Waterloo, 250 Laurelwood Dr., Waterloo, ON N2T 2S1, Canada, adrian.chee@uwaterloo.ca)

Vascular phantoms are well regarded as essential experimental tools in the development of new ultrasound techniques for assessing wall mechanics and blood flow. However, existing phantoms are ill-suited for evaluation of integrative imaging methods that seek to concurrently assess biomechanics and hemodynamics. Here, we present a novel design protocol for acoustically-

compatible anthropomorphic walled phantoms with artery-like vessel elasticity of carotid bifurcation with stenotic-narrowing. Our protocol involved a set of three-dimensional printed mold parts (consisted of a vessel core and an outer mold) for investment casting of polyvinyl alcohol solution to construct the elastic vessel tube. Agar-gelatin slab was formed around the vessel tube mimicking surrounding tissue. For demonstration, a set of healthy and stenosed (25%, 50%, and 75%) carotid bifurcation phantoms were developed. Imaging experiments were performed on these phantoms to visualize complex blood flow (recirculation and flow jet formation observed) and pulse wave dynamics (derived pulse wave velocity = 4.67 ± 0.71 m/s). Integrative imaging of wall motion and blood flow in our phantoms also revealed fluid-structure interaction differences between healthy and diseased models. These findings show that phantoms developed with our new protocol are useful in vascular imaging studies that individually or jointly assess wall motion and flow dynamics.

9:20

2aBA4. Atherosclerosis characterization using lipid-specific photoacoustic imaging and 4D ultrasound strain mapping in mice. Gurneet S. Sangha and Craig J. Goergen (Biomedical Eng., Purdue Univ., 206 S. Martin Jischke Dr., West Lafayette, IN 47907, gsangha@purdue.edu)

Dual-modality photoacoustic tomography (PAT) and four-dimensional ultrasound (4DUS) imaging have recently been used study atherosclerosis progression in small animals. PAT uses pulsed laser light to induce acoustic waves and reconstruct lipid-specific compositional images of tissue. 4DUS captures dynamic volumetric information and can be used to estimate three-dimensional (3D) Green-Lagrange strain using a direct deformation estimation method. Here, we hypothesized that PAT/4DUS can be used to correlate changes in arterial strain and hemodynamics with lipid localization and density in animals that have undergone partial carotid ligation (PCL) induced-atherosclerosis. A 40 MHz transducer (Vevo2100, VisualSonics) and a ND:YAG pulsed laser (Surelite EX, Continuum) were used to image five apolipoprotein-E deficient mice that underwent PCL of the left carotid artery while being fed a Western diet. Animals were imaged using 4DUS at days 0, 1, 4, 7, 10, and 14 to obtain pulsed-wave Doppler for hemodynamic characterization and 4DUS images for strain mapping. At day 14 all animals were euthanized and 3D *in situ* PAT images of the left carotid artery were acquired using 1210 nm light. The results show that atherosclerotic lesions can be characterized via PAT to localize both lipid accumulation and density and 4DUS to identify regions of low strain.

9:35

2aBA5. Spatial analysis of cardiac strain using high-frequency four-dimensional ultrasound in mice. Frederick W. Damen, Arvin Soepriatna, and Craig J. Goergen (Biomedical Eng., Purdue Univ., 206 S. Martin Jischke Dr., Rm. 3083, West Lafayette, IN 47906, fdamen@purdue.edu)

Cardiac disease remains the number one cause for all mortality in the United States, prompting a continued effort to understand the various factors that exacerbate disease. In this endeavor, mouse models of cardiac disease have served a crucial role by allowing for both investigation of disease factors and longitudinal tracking of cardiac function. Routine assessment of cardiac function in mice can be acquired using high-frequency ultrasound; however, conventional techniques must rely on idealized cardiac geometries to calculate function metrics, as morphometric information is only available from a single plane. Aiming to overcome these limitations, our group has recently developed and validated a high frequency four-dimensional ultrasound (4DUS) technique that provides full volumetric information of the mouse heart synced over one representative cardiac cycle. Analysis of this 4DUS data can provide region-specific wall kinematic information, in contrast to global metrics such as ejection fraction and stroke volume. Preliminary applications of our technique have demonstrated abnormal left-ventricular contractile patterns in mouse models of cardiac hypertrophy, as well as ventricular remodeling in models of myocardial infarction. These initial efforts suggest that widespread adoption of 4DUS has the potential to help increase the amount of information obtained when using mouse models of cardiac disease.

9:50

2aBA6. Quantification of murine cardiac hypertrophy using 4D ultrasound. Alycia G. Berman, Jennifer L. Anderson, Elizabeth E. Niedert (Biomedical Eng., Purdue Univ., 206 S Martin Jischke Dr., MJIS 3083, West Lafayette, IN 47907, berman1@purdue.edu), Adrienne Scott, Corey P. Neu (Mech. Eng., Univ. of Colorado at Boulder, Boulder, CO), and Craig J. Goergen (Biomedical Eng., Purdue Univ., West Lafayette, IN)

Cardiac hypertrophy is abnormal thickening, followed by dilation, of the heart which can lead to congestive heart failure. Herein, we use a mouse model of hypertrophy to explore the relationship between *in vivo* strain and the resultant hypertrophic state. To do so, osmotic pumps containing saline (n = 5) or angiotensin II (AngII; n = 10) were surgically implanted into the dorsal flank of C57BL/6J mice. AngII increased blood pressure and cardiac afterload, causing myocardial hypertrophy. Mice were imaged weekly using a VisualSonics Vevo2100 ultrasound system with a MS550D transducer (40 MHz center frequency) to collect ECG-gated Kilohertz Visualization data. In combination with a linear stepper motor, we also collected four dimensional (4D) cardiac data (3D + time). Two weeks post-surgery, pumps were removed from a subset of mice to assess the heart's ability to repair itself post-insult (n = 5). All mice were euthanized at 4 weeks. Standard metrics of left ventricular mass measured via two-dimensional slices of the 4D data indicated significantly increased mass in the AngII mice by day 14. Removal of the pump enabled significant, but partial, recovery. Current work is being performed to calculate strain within the cardiac wall. Ultimately, we aim to determine if increases in *in vivo* strain precede increases in cardiac mass.

10:05–10:20 Break

Invited Papers

10:20

2aBA7. Advanced beamforming for improved functional assessment in echocardiography. Brett Byram, Kaz Dei, Siegfried Schlunk, and Adam Luchies (Biomedical Eng., Vanderbilt Univ., 2301 Vanderbilt Pl.; PMB 351631, Nashville, TN 37235, brett.c.byram@vanderbilt.edu)

Echocardiography is one of the most used medical imaging exams. The data from these exams are often used to compute quantitative metrics of cardiac health including measures such as the global longitudinal strain (GLS) or ejection fraction. Metrics related to blood flow are also derived from echocardiography data. These metrics have great potential because ideally, they provide quantitative biomarkers to monitor cardiac function over time and compare patient function to population values. Unfortunately, echocardiography data

is often severely corrupted by various forms of acoustic clutter including wavefront aberration, reverberation and off-axis scattering from bright structures like the ribs and lungs. These sources of clutter degrade ultrasound image quality and corrupt the ability to derive reliable quantitative biomarkers of cardiac function. To resolve problems like these, we have been developing non-linear (mathematically) beamformers. These include our aperture domain model image reconstruction (ADMIRE) beamformer and, more recently, our deep neural network beamformer. We have shown that these beamformers are able to reduce errors related to wavefront aberration, reverberation and off-axis scattering, and also that the ADMIRE beamformer can eliminate the underestimation of global longitudinal strain caused by high levels of stationary reverberation caused by the chest wall.

10:40

2aBA8. Super-resolution ultrasound imaging beyond the acoustic diffraction limit. Kang Kim (Medicine, Univ. of Pittsburgh, 950 Scaife Hall, 3550 Terrace St., Pittsburgh, PA 15261, kangkim@upmc.edu), Qiyang Chen, and Jaesok Yu (BioEng., Univ. of Pittsburgh, Pittsburgh, PA)

Contrast enhanced ultrasound (CEU) imaging technologies using microbubbles (MBs) provide superior contrast of vasculatures, effectively suppressing the surrounding tissue signals, but the spatial resolution remains to the acoustic diffraction limit. By localizing the center of each MBs unprecedented high spatial resolution beyond the acoustic diffraction limit can be achieved. However, some methods of localizing each center of the signals from individual MBs that only applies symmetrically distributed signal amplitude require a large number of imaging frames, especially when MBs are densely clumped, therefore result in a long scan time that is not ideal for *in vivo* scan under physiologic conditions. In this paper, we present an innovative approach using deconvolution technique that will allow for identifying signals from individual MBs from dense population in any forms even grouped together within the full-width-at-half-maximum (FWHM) of the point spread function (PSF) of the US probe. In this way, no collected frame sets require to be excluded for image reconstruction, therefore scan time can be reduced significantly. *In vivo* application of this new approach in identifying vasa vasorum in rabbit atherosclerotic plaque model will be presented. Some technical limitations including background noise as well as motion artifact will be discussed.

Contributed Papers

11:00

2aBA9. Motion-resistant vascular ultrasound imaging based on real-time eigen-filtering. Adrian J. Chee, Billy Y. Yiu, and Alfred C. Yu (Univ. of Waterloo, EIT 4125, Waterloo, ON N2L 3G1, Canada, alfred.yu@uwaterloo.ca)

Doppler flow imaging has become a standard clinical modality for vascular diagnostics. Nevertheless, it remains challenging to perform vascular ultrasound in more complicated diagnostic scenarios, because significant flashing artifacts may appear due to inadequate suppression of Doppler clutter arising from moving tissues. For one decade, we have been striving to achieve motion-resistant vascular ultrasound by designing advanced eigen-filtering algorithms whose attenuation response is adapted to clutter characteristics. Using a receiver operating characteristics analysis approach, we showed that in the presence of vessel pulsation and tissue vibration, our eigen-based motion-resistant signal processing chain yielded a significantly higher true positive rate (>90%) in depicting flow in comparison to non-adaptive signal processing chains. Another engineering challenge that we have overcome is the high computational demand of eigen-processing algorithms. We have successfully devised real-time implementations of eigen-based motion-resistant signal processing through designing parallel computing kernels that are executed on a graphical processing unit (GTX Titan X). In particular, we achieved real-time video-range throughput for full-view Doppler frames, up to a scan depth of 5 cm for slow-time ensemble length of 16 samples (i.e., beyond the typical requirement for carotid scans). These findings serve well to substantiate the practical feasibility of performing motion-resistant vascular ultrasound.

11:15

2aBA10. Ascertaining the relationship between acoustic droplet vaporization, inertial cavitation, and hemolysis. Newsha Jahanpanah (Medical Sci. Baccalaureate Program, Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, jahanpna@mail.uc.edu), Sneha Sharma, Karla P. Mercado-Shekhar, Haili Su (Div. of Cardiovascular Health and Disease, Univ. of Cincinnati, Cincinnati, OH), Hunter A. Palcich, Austin M. Wanek (Dept. of Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), and Kevin J. Haworth (Div. of Cardiovascular Health and Disease, Univ. of Cincinnati, Cincinnati, OH)

Mechanical stress associated with inertial cavitation (IC) can cause hemolysis. The goal of this study was to determine whether hemolysis occurs during acoustic droplet vaporization (ADV) and subsequent microbubble activity. Perfluoropentane droplets were manufactured using an amalgamation protocol, diluted 30-fold in porcine whole blood, and pumped through a flow phantom maintained at 37 °C. Droplets were insonified with a 5 MHz focused transducer (peak negative pressures of 0.25 MPa to 4.25 MPa, pulse length of 40 cycles, and a pulse repetition frequency of 500 Hz). A 128-element array was used to monitor the nucleation of ADV and to record cavitation emissions. Free hemoglobin in the effluents was assayed to detect hemolytic activity. Hemolysis, IC, and ADV were detected at pressures of 2.25 MPa and higher. At 4.25 MPa, 0.38 ± 0.08% hemoglobin was released post-hemolysis. A positive linear relationship was observed between the percent hemoglobin released and IC dose. Due to the concurrence of ADV and IC, it was not possible to determine whether ADV causes hemolysis independent of IC. These findings should be considered when planning studies of biomedical application of ADV. [Work supported in part by an Acoustic Society of America Robert W. Young Award.]

11:30

2aBA11. Frequency dependence of the vaporization threshold of sonosensitive perfluorocarbon droplets varying their liquid core and size. Mitra Aliabouzar (George Washington Univ., 8222 Harvest Bend Ln. APT. #37, Laurel, MD 20707, mitraaali@email.gwu.edu), Krishna Kumar (George Washington Univ., Washington, DC), and Kausik Sarkar (George Washington Univ., Washington, DC)

Phase shift liquid perfluorocarbon (PFC) droplets vaporizable by ultrasound have received increasing attention for therapeutic and diagnostic applications. The ultrasound activation pressure required for the phase change of these droplets into echogenic microbubbles is termed acoustic droplet vaporization (ADV). This study systematically investigates the effect of frequency of excitation¾2.25 MHz, 10 MHz and 15 MHz¾on ADV and inertial cavitation (IC) thresholds of a suspension of PFC droplets by varying the physical properties of the liquid droplets (size and boiling point) in a tubeless setup. We prepared lipid-coated droplets with three different liquid cores, perfluoropentane (PFP), perfluorohexane (PFH) and perfluorooctyle bromide (PFOB), of two different size ranges ¾ one with diameter smaller than 3 ö¥m and the other with diameter larger than 10 ö¥m. We found that the ADV threshold increases with frequency for the most volatile PFC liquid, PFP, for both large and small size droplets. While for the small size droplets of higher boiling point liquids, PFH and PFOB, no ADV was detected, for the larger ones ADV threshold decreases with frequency of excitation. ADV thresholds at all the frequencies studied here

occurred at lower rarefactional pressures than IC thresholds indicating that phase transition precedes inertial cavitation.

11:45

2aBA12. Acoustic droplet vaporization with microfluidic droplets results in dissolved oxygen scavenging. Rachel P. Benton (Neurosci. Baccalaureate Program, Univ. of Cincinnati, 213 River Valley Blvd., Cincinnati, OH 45157, bentonrp@mail.uc.edu), Newsha Jahanpanah (Medical Sci. Baccalaureate Program, Univ. of Cincinnati, Cincinnati, OH), Emilee Warner (Dept. of Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Rohan S. Srivastava (Medical Sci. Baccalaureate Program, Univ. of Cincinnati, Cincinnati, OH), Ishan Anand (Dept. of Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), and Kevin J. Haworth (Div. of Cardiovascular Health and Disease, Univ. of Cincinnati, Cincinnati, OH)

Acoustic droplet vaporization (ADV) is the ultrasound-mediated phase transitioning of liquid perfluoropentane (PFP) droplets into gas microbubbles resulting in dissolved oxygen scavenging from the surrounding fluid. The objective of this study was to determine how droplet diameter influences oxygen scavenging. Droplets of 12 μm and 6 μm diameters were manufactured using a microfluidic system, which were diluted in saline to obtain concentrations of 0.48 X 10^{-3} ð 0.12 X 10^{-3} ml/ml and 1.11 X 10^{-3} ð 0.13 X 10^{-3}ml/ml, respectively. Samples were pumped through a 37 °C flow phantom at 10 ml/min. A 5 MHz transducer insonified droplets at 5 MPa for 20 cycles with a 500 Hz repetition frequency. Oxygen partial pressure (Po_2) was measured with a distal sensor. Samples of 12 μm and 6 μm droplets had an average volume-weighted transition efficiency of 5.8% $\pm$ 4.5% and 5.3% $\pm$ 4.6%, respectively. The initial Po_2 was 169 $\pm$ 7 mmHg for all samples. ADV with 12 μm and 6 μm droplets reduced the Po_2 to 113 $\pm$ 21 mmHg and 101 $\pm$ 9 mmHg, respectively. An oxygen scavenging model based on the PFP phase transitioned volume yielded a final Po_2 of 121 $\pm$ 27 mmHg for 12 μm droplets and 106 $\pm$ 29 mmHg for 6 μm droplets. There was no statistically significant difference ($p > 0.05$) between the experimentally measured and modeled Po_2 values after ADV.

TUESDAY MORNING, 14 MAY 2019 — EXHIBIT HALL, 8:00 A.M. TO 11:30 A.M.

Session 2aID

Interdisciplinary and Student Council: Graduate Programs in Acoustics (Poster Session)

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, PA 16802

Michael Rollins, Cochair
Biomedical Engineering, University of Cincinnati, MSB 6303, 231 Albert Sabin Way, Cincinnati, OH 45267

Matthew C. Zeh, Cochair
Mechanical Engineering Graduate Acoustics Program, University of Texas at Austin, 3374 Lake Austin Blvd., Apt. B, Austin, TX 78703

All posters will be on display from 8:00 a.m. to 11:30 a.m. To give contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 9:45 a.m. and contributors of even-numbered papers will be at their posters from 9:45 a.m. to 11:30 a.m.

Invited Papers

2aID1. Graduate acoustics at Brigham Young University. Tracianne B. Neilsen, Scott D. Sommerfeldt, Timothy W. Leishman, Kent L. Gee, Brian E. Anderson, Jonathan Blotter, Scott L. Thomson, and William Strong (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu)

Graduate studies in acoustics at Brigham Young University prepare students for industry, research, and academia by complementing in-depth coursework with publishable research. Graduate-level coursework provides students with a solid foundation in core acoustics principles and practices and measurement skills, including a strong foundation in experimental techniques and writing technical memoranda. Labs across the curriculum cover calibration, directivity, scattering, absorption, laser Doppler vibrometry, lumped-element mechanical systems, equivalent circuit modeling, arrays, filters, room acoustics, active noise control, and near-field acoustical holography. Recent thesis and dissertation topics include active noise control, directivity of acoustic sources, room acoustics, radiation and directivity of musical instruments, energy-based acoustics, time reversal, nondestructive evaluation, flow-based acoustics, voice production, aeroacoustics, sound propagation modeling, nonlinear propagation, high-amplitude noise analyses, machine and deep learning applied to ambient noise level prediction, crowd noise interpretation, underwater acoustic source localization, and ocean environment classification. The graduate students are expected to present research at professional meetings and publish in peer-reviewed acoustics journals. Graduate students often serve as peer mentors to undergraduate students on related projects and may participate in field experiments to gain additional experience. For update, follow us @BYUAcoustics

2aID2. Graduate programs in acoustics at the University at Buffalo, State University of New York. Anastasiya Kobrina and Kali Burke (Psych., SUNY Univ. at Buffalo, B23 Park Hall, Amherst, NY 14261, akobrina@buffalo.edu)

University at Buffalo is known for its diversity in acoustic and auditory research spanning from signal processing to hearing in humans and animals. These research programs are nested within Anthropology, Biological Sciences, Communicative Disorders and Sciences, Computer Science and Engineering, Linguistics, Psychology, and Music, which leads to a unique variety of collaborations spanning departments and laboratories. The doctoral programs in these departments are aimed at training students for clinical and research-oriented careers. Our graduate programs involve taking extensive coursework and hands on experience in laboratories. In addition, each department holds regular colloquia on various topics in acoustics. University at Buffalo is also home to the Center for Hearing and Deafness. The Center seeks to develop cooperative working relationships with businesses and industries involved in hearing-related activities, such as: hosting the WNY Tinnitus Support Group, testing and evaluating drugs used to treat hearing loss, developing new scientific and clinical instrumentation, and assessing industrial hearing loss and noise regulations. The Center also provides valuable training opportunities for physicians, engineers, and health professionals. In conclusion, the University at Buffalo is an ideal fit for training in acoustics research.

2aID3. Biomedical acoustics research at the Image-Guided Ultrasound Therapeutics Laboratories. Christy K. Holland, T. Douglas Mast, and Kevin J. Haworth (Internal Medicine, Div. of Cardiovascular Health and Disease, and Biomedical Eng., Univ. of Cincinnati, Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

The Image-guided Ultrasound Therapeutic Laboratories (IgUTL) are located at the University of Cincinnati in the Heart, Lung, and Vascular Institute, a key component of efforts to align the UC College of Medicine and UC Health research, education, and clinical programs. These laboratories, directed by Christy K. Holland, comprise graduate and undergraduate students, postdoctoral fellows, physician-scientists, and clinical and scientific collaborators in fields including cardiology, neurosurgery, neurology, radiology, and emergency medicine. Holland's research focuses on biomedical ultrasound including sonothrombolysis, ultrasound-mediated drug and bioactive gas delivery, development of echogenic liposomes, early detection of cardiovascular diseases, and ultrasound-image guided tissue ablation. Imaging algorithms incorporate both passive and active cavitation detection. The Biomedical Acoustics Laboratory within IgUTL, directed by T. Douglas Mast, investigates ultrasound imaging for therapy guidance, including echo decorrelation imaging for monitoring liver cancer ablation and real-time tracking of tongue motion for biofeedback in speech therapy. The Biomedical Ultrasonics and Cavitation Laboratory within IgUTL, directed by Kevin J. Haworth, employs ultrasound-mediated gas scavenging for image-guided treatment of cardiovascular disease, especially reperfusion injury. [Work supported by NIH Grants R01 NS047603, R01 HL135092, R01 HL133334, R01 CA158439, R01 DC017301, and K25 HL133452.]

2aID4. Graduate studies in acoustics and wave physics at Institut d'Acoustique—Graduate School, Le Mans, France. Vincent Tournat (LAUM, CNRS UMR 6613, Le Mans Université, Av. O. Messiaen, Av. O. Messiaen, Le Mans 72085, France, vincent.tournat@univ-lemans.fr)

This poster presents the graduate studies in Acoustics at Le Mans (France) offered at the Institut d'Acoustique—Graduate School. Graduate studies in Acoustics at Le Mans University have been awarded in 2018 the excellence label École Universitaire de Recherche among 28 other reference centers for all fields of Science, through a highly selective national call. Master and engineering school programs range from physical acoustics, environmental acoustics, acoustics and vibrations to international masters on electro-acoustics and on wave physics. The education through research is carried out at the LAUM, UMR CNRS, one of the largest acoustics laboratory in the world. Several details, objectives and contact informations on the graduates studies will be given on the poster.

2aID5. Graduate training opportunities in the hearing sciences at the University of Louisville. Shae D. Morgan (Dept. of Otolaryngol. and Communicative Disord., Univ. of Louisville, 627 S. Preston St., Ste. 220, Louisville, KY 40292, shae.morgan@louisville.edu), Hammam AlMakadma (Dept. of Otolaryngol. and Communicative Disord., Univ. of Louisville, Syracuse, New York), Maria V. Kondaurova, Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY), and Pavel Zahorik (Dept. of Otolaryngol. and Communicative Disord., Heuser Hearing Inst. and Univ. of Louisville, Louisville, KY)

The University of Louisville currently offers two branches of training opportunities for students interested in pursuing graduate training in the hearing sciences: A Ph.D. degree in experimental psychology with concentration in hearing science, and a clinical doctorate in audiology (Au.D.). The Ph.D. degree program offers mentored research training in areas such as psychoacoustics, speech perception, spatial hearing, and multisensory perception, and guarantees students four years of funding (tuition plus stipend). The Au.D. program is a 4-year program designed to provide students with the academic and clinical background necessary to enter audiologic practice. Both programs are affiliated with the Heuser Hearing Institute, which, along with the University of Louisville, provides laboratory facilities and clinical populations for both research and training. An accelerated Au.D./Ph.D. training program that integrates key components of both programs for training of students interested in clinically-based research is under development. Additional information is available at http://louisville.edu/medicine/degrees/audiology and http://louisville.edu/psychology/graduate/vision-hearing.

2aID6. Graduate programs at the University of Maryland. Eric C. Hoover, Samira B. Anderson, Matthew Goupell, and Sandra Gordon-Salant (Dept. of Hearing and Speech Sci., Univ. of Maryland, 7251 Preinkert Dr., College Park, MD 20742, ehoover@umd.edu)

The University of Maryland has many opportunities for graduate studies in topics in acoustics, including hearing, speech and language sciences, neuroscience, physiology, bioacoustics, linguistics, and acoustical and communications engineering. Graduate training at the University of Maryland is characterized by an emphasis on multidisciplinary, collaborative research designed to enable students to develop a breadth of knowledge in addition to their focused research program. There are numerous interdisciplinary initiatives that address problems of great importance to science and society. In addition to the interdisciplinary research training on campus, students also train and collaborate with world-class researchers from nearby institutions including National Institutes of Health, Walter Reed National Military Medical Center, the University of Maryland Medical Center, and many other outstanding institutions in the area. Individuals interested in graduate training at the University of Maryland should contact faculty in their field of interest to learn more about the many opportunities available.

2aID7. Graduate studies in Acoustical Oceanography in the Massachusetts Institute of Technology and Woods Hole Oceanographic Institution Joint Program. Andone C. Lavery (Woods Hole Oceanographic Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu)

An overview of graduate studies in Acoustical Oceanography within the framework of the Massachusetts Institute of Technology (MIT) and Woods Hole Oceanographic Institution (WHOI) Joint Program is presented, including a brief history of the program, facilities, details of the courses offered, alumni placing, funding opportunities, and current program status, faculty members and research. Emphasis is given to the key role of the joint strengths provided by MIT and WHOI, the strong sea-going history of the program, and the potential for highly interdisciplinary research.

2aID8. Graduate research opportunities in acoustics at the University of Michigan, Ann Arbor. Tyler J. Flynn and David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, tjayflyn@umich.edu)

The University of Michigan (UM) is host to a wide array of acoustics research which encompasses many of the Technical Committees of the ASA. Within the Department of Mechanical Engineering alone, work is being done to develop better remote sensing techniques for underwater environments, to better understand the mechanics of the human cochlea, and to build metamaterials allowing for new and exotic acoustic behaviors. Within the UM Medical School, faculty and graduate students are constantly advancing techniques for diagnostic and therapeutic ultrasound procedures. In the Department of Naval Architecture and Marine Engineering, computational and experimental tools are being developed to enable better ship design. In Electrical Engineering, advances in MEMS fabrication yield even smaller and more efficient transducers for smarter devices. And researchers in the Linguistics Department are using the fundamental acoustic processes of speech to learn how humans effectively communicate. And while these are only a sample of the projects taking place at Michigan, new opportunities for acoustics research and collaboration open up each semester. Combined with a rich course catalogue, first-rate facilities, and great prospects for publication, these opportunities prepare UM graduate students for careers in industry and academia alike. Go Blue!

2aID9. Graduate study in physical acoustics at the University of Mississippi. Joel Mobley, Josh R. Gladden, Cecille Labuda (Phys. and Astronomy, Univ. of Mississippi, P.O. Box 1848, 1034 NCPA, University, MS 38677, jmobley@olemiss.edu), and Likun Zhang (Phys. and Astronomy, Univ. of Mississippi, Oxford, MS)

The University of Mississippi is a Ph.D. granting institution with an R1 Carnegie designation placing it among schools with the highest level of research activity. The Department of Physics and Astronomy at Ole Miss has a diverse range of research opportunities, including two groups associated with recent Nobel Prizes. Along with programs in Computational Physics, High Energy Physics, Atmospheric Physics and Gravitation, the department is affiliated with the National Center for Physical Acoustics (NCPA). NCPA is an 85 000 square foot standalone facility on the campus of the University of Mississippi dedicated to the physics and engineering applications of acoustics. It has research groups dedicated to ultrasound, infrasound, aeroacoustics, atmospheric propagation, porous media, and ocean acoustics. Graduate students in physics and engineering are pursuing Ph.D. and M.S. degrees at NCPA, and four faculty members from the Physics department have their research laboratories in the facility. In addition to NCPA, the Physics department has affiliations with the Laser Interferometer Gravitational Wave Observatory (LIGO), The European Center for Particle Physics (CERN), Fermilab and Belle II.

2aID10. Opportunities for graduate studies in acoustics within the College of Engineering at the University of Nebraska–Lincoln. Lily M. Wang, Erica E. Ryherd (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu), Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska - Lincoln, Lincoln, NE), and Jinying Zhu (Civil Eng., Univ. of Nebraska - Lincoln, Omaha, NE)

A number of faculty in the College of Engineering at the University of Nebraska–Lincoln (UNL) conduct research in acoustics and mentor graduate students. Within the Durham School of Architectural Engineering and Construction, based at UNL's Scott Campus in Omaha, Lily Wang and Erica Ryherd are active in architectural acoustics and noise (http://nebraskaacousticsgroup.org). In Civil Engineering, Jinying Zhu (also based on UNL's Scott Campus in Omaha) focuses in structural acoustics, using ultrasonic waves for concrete evaluation. In Mechanical and Materials Engineering, Joseph Turner (based at UNL's City Campus in Lincoln) studies ultrasound propagation through complex media for quantitative characterization of materials/microstructure (http://quisp.unl.edu). UNL is additionally home to an active student chapter of the Acoustical Society of America, the first to be founded in 2004. This poster will summarize the graduate-level acoustics courses and lab facilities at UNL within the College of Engineering, as well as the research interests and achievements of our faculty, graduates, and students.

2aID11. Graduate studies in acoustics at the University of New Hampshire. Daniel R. Howard, Anthony P. Lyons (Univ. of New Hampshire, Durham, NH), Jennifer L. Miksis-Olds (Univ. of New Hampshire, Durham, North Carolina), and Thomas C. Weber (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, tom.weber@unh.edu)

The University of New Hampshire (UNH) offers several opportunities for graduate students interested in studying acoustics and its application. Faculty mentors who are expert in acoustic methods and technologies reside in a range of programs and departments that are largely focused on the use of acoustics in the marine environment, including biological science, earth science, mechanical engineering, natural resources and earth systems, ocean engineering, and oceanography. UNH faculty mentors who specialize in acoustics are active in the Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics technical committees. Recent studies by faculty and students focusing on fundamental acoustic problems, such as those that would cause a graduate student to be a regular attendee of meetings of the Acoustical Society of America, have come largely from mechanical engineering, ocean engineering, biological sciences, and the newly formed School of Marine Sciences and Ocean Engineering. Graduate students in these programs of study have the opportunity for formal classroom training in the fundamentals of acoustics, vibrations, and advanced topics in acoustics as they pursue their graduate training.

2aID12. Graduate acoustics research at the University of New Haven. Eric A. Dieckman (Mech. Eng., Univ. of New Haven, 300 Boston Post Rd., West Haven, CT 06516, edieckman@newhaven.edu)

The University of New Haven offers graduate study opportunities in acoustics, primarily through a Master of Science in Mechanical Engineering degree. This flexible program allows students to tailor a curriculum toward their interests in preparation for further graduate studies or entry into the workforce. Courses offered include Fundamentals of Acoustics, System Vibrations, Nondestructive Evaluation, and Wave Propagation and Scattering. All classes have a focus on hands-on projects. A one-semester project or two-semester thesis allows students structured time to work on research problems in acoustics.

2aID13. Pursuing graduate degrees in acoustics at Penn State. Victor Sparrow and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dar119@psu.edu)

The Graduate Program in Acoustics at Penn State is the only program in the U.S. offering the Ph.D. in Acoustics as well as M.S. and M.Eng. degrees in Acoustics. An interdisciplinary program with faculty from a variety of academic disciplines, the Graduate Program in Acoustics is administratively aligned with the College of Engineering and closely affiliates with the Applied Research Laboratory. Research areas include: structural acoustics, nonlinear acoustics, architectural acoustics, signal processing, aeroacoustics, biomedical ultrasound, transducers, computational acoustics, noise and vibration control, psychoacoustics, and underwater acoustics. Course offerings include fundamentals of acoustics and vibration, electroacoustic transducers, signal processing, acoustics in fluid media, sound and structure interaction, digital signal processing, experimental techniques, acoustic measurements and data analysis, ocean acoustics, architectural acoustics, noise control engineering, nonlinear acoustics, outdoor sound propagation, computational acoustics, biomedical ultrasound, flow induced noise, spatial sound and three-dimensional audio, acoustics of musical instruments. More than 700 Penn State Acoustics graduates serve widely throughout military and government labs, academic institutions, consulting firms and industry. This poster describes faculty research areas, laboratory facilities, student demographics, successful graduates, and recent enrollment and employment trends.

2aID14. A master's degree in acoustics through distance education from Penn State. Daniel A. Russell and Victor Sparrow (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dar119@psu.edu)

The Graduate Program in Acoustics at Penn State provides online access to graduate level courses leading to the M.Eng. degree in Acoustics. Lectures are broadcast live via Adobe Connect to students scattered around the world, while archived recordings allow working students to access lectures at their convenience. Students earn the M.Eng. in Acoustics degree by completing 30 credits of coursework (six required courses and four electives) and writing a capstone paper. Since 1987, more than 150 distance education students have completed the M.Eng. in Acoustics degree. Many other students take individual courses as non-degree students. Courses offered online include: elements of acoustics and vibration, elements of waves in fluids, electroacoustic transducers, signal processing, acoustics in fluid media, sound and structure interaction, digital signal processing, aerodynamic noise, acoustic measurements and data analysis, ocean acoustics, architectural acoustics, noise control engineering, nonlinear acoustics, outdoor sound propagation, computational acoustics, biomedical ultrasound, flow induced noise, spatial sound and 3D audio, acoustics of musical instruments. This poster describes the distance education experience leading to the M.Eng. degree in Acoustics from Penn State and showcases student demographics, capstone paper topics, enrollment statistics and trends, and the success of our graduates.

2aID15. Underwater acoustics and ocean engineering at the University of Rhode Island. Lora J. Van Uffelen, James H Miller, and Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., 213 Sheets Lab., Narragansett, RI 02882, loravu@uri.edu)

Underwater acoustics is one of the primary areas of emphasis in the Ocean Engineering Department at the University of Rhode Island, the first Ocean Engineering program in the United States. The program offers Bachelors, Masters (thesis and non-thesis options) and Ph.D. degrees in Ocean Engineering. These programs are based at the Narragansett Bay campus, providing access to a living laboratory for student learning. Some key facilities of the program are an acoustics tank, a 100-foot-long wave tank, and currently the R/V Endeavor, a UNOLS oceanographic research vessel operated by the University of Rhode Island. A new Regional Class vessel is anticipated in 2021. At the graduate level, students are actively involved in research focused in areas such as acoustical oceanography, propagation modeling, acoustic positioning and navigation, geoacoustic inversion, marine mammal acoustics, ocean acoustic instrumentation, and transducers. An overview of classroom learning and ongoing research will be provided, along with information regarding the requirements of entry into the program.

2aID16. Graduate acoustics education in the Cockrell School of Engineering at The University of Texas at Austin. Michael R. Haberman, Mark F. Hamilton, Preston S. Wilson (Walker Dept. Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu), Neal A. Hall (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

While graduate study in acoustics takes place in several colleges and schools at The University of Texas at Austin (UT Austin), including Communication, Fine Arts, Geosciences, and Natural Sciences, this poster focuses on the acoustics program in the Cockrell School of Engineering. The core of this program resides in the Walker Department of Mechanical Engineering (ME) and the Department of Electrical and Computer Engineering (ECE). Acoustics faculty in each department supervise graduate students in both departments. One undergraduate and eight graduate acoustics courses are cross-listed in ME and ECE. Instructors for these courses include staff at Applied Research Laboratories at UT Austin, where many of the graduate students have research assistantships. The undergraduate course, taught every fall, begins with basic physical acoustics and proceeds to draw examples from different areas of engineering acoustics. Three of the graduate courses are taught every year: a two-course sequence on physical acoustics, and a transducers course. The remaining five graduate acoustics courses, taught in alternate years, are on nonlinear acoustics, underwater acoustics, ultrasonics, architectural acoustics, and wave phenomena. An acoustics seminar is held most Fridays during the long semesters, averaging over ten per semester since 1984. The ME and ECE departments both offer Ph.D. qualifying exams in acoustics.

2aID17. Graduate education and research in architectural acoustics at Rensselaer Polytechnic Institute. Ning Xiang, Jonas Braasch, and Todd Brooks (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu)

The Graduate Program in Architectural Acoustics has been constantly advanced from its inception in 1998 with an ambitious mission of educating future experts and leaders in architectural acoustics, due to the rapid pace of change in the fields of architectural-, physical-, and psycho-acoustics. Recent years the program's pedagogy using science, technology, engineering, and mathematics methods (STEM) has been proven to be effective and productive, including intensive, integrative hands-on experimental components that integrate architectural acoustics theory and practice. The graduate program has recruited graduate students from a variety of disciplines including individuals with B.S., B.Arch., or B.A. degrees in Engineering, Physics, Mathematics, Architecture, Electronic Media, Sound Recording, Music and related fields. Under this pedagogy and research environment graduate students have been succeed in the rapidly changing field. RPI's Graduate Program in Architectural Acoustics has since graduated more than 150 graduates with both M.S. and Ph.D. degrees. Under the guidance of the faculty members they have also actively contributed to the program's research in architectural acoustics, noise control, psychoaoustics, signal processing in acoustics as well as our scientific exploration at the intersection of cutting edge research and traditional architecture / music culture. This paper illuminates the evolution and growth of the graduate program.

2aID18. Opportunities for graduate studies in physical, engineering, and underwater acoustics at the Naval Postgraduate School. Oleg A. Godin, Kevin B. Smith, and Anne T. Pickens (Phys. Dept., Naval Postgrad. School, 33 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu)

The Departments of Physics and of Electrical and Computer Engineering at the Naval Postgraduate School offer programs of graduate studies in acoustics leading to MS degrees in physics, applied physics, and engineering acoustics and PhD degrees in engineering acoustics and applied physics. The program is interdisciplinary, with courses and laboratory work drawn principally from the fields of physics and electrical engineering. Subjects covered include waves and oscillations; fundamentals of physical and structural acoustics; the generation, propagation and reception of sound in the ocean; civilian and military applications of sonar systems; and acoustic signal processing. Topics of recent theses and dissertations include development and field testing of novel sensors for atmospheric and ocean acoustics, modeling and measurements of ambient noise and sound propagation in the ocean, sound scattering in underwater waveguides, acoustic vector sensors and vector field properties, acoustic communications, geo-acoustic inversion, acoustic remote sensing of the ocean, and acoustics of autonomous underwater and aerial vehicles.

TUESDAY MORNING, 14 MAY 2019 — BREATHITT, 9:00 A.M. TO 11:45 A.M.

Session 2aMU

Musical Acoustics: General Topics in Musical Acoustics

Randy Worland, Chair
Physics, University of Puget Sound, 1500 N. Warner, Tacoma, WA 98416

Contributed Papers

9:00

2aMU1. Non-destructive correlation of Nigerian drum beat-pattern and pitch to detect a ripen watermelon. Stephen G. Onwubiko (Music, Univ. of Nigeria, Nsukka Enugu State, Enugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com) and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

In the application of musical acoustics and speech sound, almost any type of Nigerian drum is used for communication. The agreeable successions of tones unlimited to interesting beat-patterns, pitch and rhythms uses shifted accents, non-accented rhythms, syncopations etc. On the daily basis of a watermelon harvesting and trade point, musical acoustics and speech sounds are applied passively in detecting and determining a ripen watermelon; the application of the Nigerian drum beat-pattern, pitch and intonation is an efficient procedure for ripeness detection of watermelon. Depending on how the pitches are lowered or accented, the melon ripeness is detected. The pitch-pattern analysis can be used to measure, determine and correlate the internal ripeness and quality of watermelon with pitch from a Nigerian drum. This method allows identification at a 60.0 % level of efficiency. Hence, the proposed method can reliably detect watermelon ripeness.

9:15

2aMU2. The effect of axial vibrations on the input impedance of the trumpet. Brooke Rodgers and Thomas Moore (Dept. of Phys., Rollins College, Box 2743, Winter Park, FL 32789, brodgers@rollins.edu)

It has been shown that the vibrations in the bell section of the trumpet can influence the sound produced by the instrument. Recent research has indicated that these effects may be traced the axial vibrations of the bell. We report experimental results that relate the input impedance of the trumpet to the phase of externally induced longitudinal motion. These results indicate that the phase difference between the driver and the bell motion can significantly affect the input impedance. [Work supported by NSF Grant #PHY-160749.]

9:30

2aMU3. Nonlinear generation of sum frequencies in Sitka spruce. Jade Case, Lauren Neldner, and Thomas Moore (Dept. of Phys., Rollins College, Box 2743, Winter Park, FL 32789, jcase@rollins.edu)

It has been known for several decades that there are anomalous frequency components in the sound of a piano. These occur at the sum and difference frequencies of the overtones attributable to string motion, and are commonly referred to as phantom partials. Recent work has shown that, contrary to the generally accepted theory, the majority of power in these frequency components originates in the wooden parts of the piano and not the string. However, the etiology of the nonlinearity is not clear. To determine if the source of the nonlinearity is the stress caused by the pressure of the strings on the soundboard, experiments were performed on isolated pieces of Sitka spruce. The results indicate that stress can significantly increase the nonlinear response of the wood. [Work supported by NSF Grant #PHY-160749.]

9:45

2aMU4. High frequency cavity modes of a Helmholtz resonator excited by an air jet. Emma Shaw (Phys., Agnes Scott College, 141 E. College Ave., Decatur, GA 30030, eshaw@agnesscott.edu), James P. Cottingham, and Robert Stills (Phys., Coe College, Cedar Rapids, IA)

In 1990, Khosropour and Millet reported on a study of the internal spectrum of a Helmholtz resonator excited by an air jet and reported that the data for frequency and amplitude of the Helmholtz mode as a function of jet speed show a series of domains separated by narrow transition regions [*JASA* **88**, 1211–1221 (1990)]. The idea was to observe the behavior of the air jet when the resonator effectively has just a single mode, and does not overblow to higher frequency modes at moderate jet velocities. Further research explored the effects of changes in neck length and jet angle, but higher resonator modes were ignored. This paper presents characterizations of some of the higher modes of cavity vibration, which can be excited at high jet velocities. Unlike the Helmholtz mode, these modes depend on the shape of the cavity shape as well as the volume. Several of these modes were observed, sometimes with more than one mode sounding simultaneously. Guided by finite element simulations, the nature of these higher cavity modes was verified experimentally by observing frequencies and locating nodal surfaces.

10:00–10:15 Break

10:15

2aMU5. Comparative study of oboe and clarinet. Laura Fitzgerald and Gordon P. Ramsey (Loyola Univ. Chicago, 1032 W Sheridan Rd., Chicago, IL 60660, lfitzgerald1@luc.edu)

The oboe and clarinet are relatively similar instruments in their size and shape yet create such different sounds. This study is an exploration of the reasoning behind their differences through a detailed acoustical analysis and geometrical study of these instruments, comparing and contrasting their properties. The oboe and clarinet are comparable in size but have some key differences. The oboe's conical shape allows for all the harmonics to be present, while the clarinet is mostly cylindrical, except for the bell, so that mostly odd harmonics are present. The oboe's keys have holes much smaller than those of the clarinet. Looking to find differences in acoustical spectra based on their geometric differences, samples of low, medium, and high ranges on both instruments using the same concert pitches have been taken. Data have been taken in a regular lab setting as well as an anechoic chamber in order to look for differences in harmonic content for each instrument. The data are consistent with the notion that the shape of the instruments does contribute to the spectra.

10:30

2aMU6. Control of vocal loudness in singing. Ingo R. Titze (National Ctr. for Voice and Speech, Univ. of Utah, 136 South Main St., Ste. 320, Salt Lake City, UT 84101-3306, ingo.titze@utah.edu)

Vocal loudness in sones is quantified on the basis of changes in spectral slope and harmonic tuning in a range of singing fundamental frequencies of 125 Hz to 1000 Hz. Spectral slope of the mouth output pressure is bracketed in the range of -3 dB/octave to -12 dB/octave to reflect a typical glottal spectrum from breathy to pressed adduction. To approximate formant tuning of harmonics, the SPL level of the first three harmonics is raised by 10, 20, and 30 dB. It is shown that spectral slope change is more effective in increasing vocal loudness than tuning a single harmonic with a vocal tract resonance. Some applications to amplified and unamplified vocal production and respective training are given.

10:45

2aMU7. Feedback control of acoustic musical instruments when the number of sensors and actuators differ. Edgar J. Berdahl (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu)

Actuated instruments can be created in a variety of ways. An interesting class of actuated instruments are feedback-controlled acoustic musical instruments. A robust way to create these is to employ pairs of collocated sensors and actuators, and then to use the feedback control to simulate virtual physical systems. In the linear and time-invariant case, this means that the feedback control functions can be designed to be positive real. Accordingly, the physical properties of the instrument can become adjustable via the feedback control. An interesting case arises when the number of actuators and sensors differ. The actuators and sensors can however still potentially be collocated, which will result in the individual transfer functions from the actuators to the sensors (e.g. the mobilities) in being positive real. Under some special cases, stable feedback control can still be attained for a wide variety of feedback gains. The Feedback Guitar serves as an interesting case study for this. It has one actuator, which is approximately collocated with six piezoelectric sensors, one for each string. Using any non-negative linear combination of the sensor signals, an approximately positive real mobility can be obtained, which can enable stable feedback control for a wide variety of feedback gains.

11:00

2aMU8. Metamaterials in musical acoustics: A modified frame drum. Rolf Bader (Inst. of Systematic Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de), Jost L. Fischer, Malte Münster, and Patrick Kontopidis (Inst. of Systematic Musicology, Univ. of Hamburg, Hamburg, Germany)

Acoustic metamaterials have properties not found in nature, like negative stiffness, cloaking behavior or frequency bandgap damping. This is achieved by complex geometries often constructed out of multiple subunits in sub-wavelength size. Although also musical instruments often have complex shapes, like guitar or piano soundboards with regular fan bracing, metamaterials have not explicitly been used here. As an example, a modified frame drum is proposed with increased sound possibilities by adding masses to the drum membrane arranged in a circle. Such structures have been shown to have cloaking behavior. Using microphone-array and laser interferometry measurements it is shown that such a drum has a frequency-dependent cloaking behavior. When struck at the center of the added circle most energy above about 400 Hz stays in the circle and decays strongly. Such a sound cannot be produced with a regular frame drum. When struck outside the circle the drum sounds very much like a regular drum without added masses. By gradually changing the playing position from the circle center towards the circle rim, frequencies above about 400 Hz are gradually added. Therefore such a modified frame drum has much more possible sounds and therefore ways of musical articulation.

11:15

2aMU9. Automatic transcription of solo audio into music notation. Dong Hyun Lee (Dept. of Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL 61801, dlee134@illinois.edu) and James W. Beauchamp (Dept. of Elec. and Comput. Eng., School of Music, Univ. of Illinois Urbana-Champaign, Urbana, IL)

Creating sheet music can be an arduous task for instrumentalists and vocalists who improvise. This problem can be solved by using a pitch detector that produces a MIDI data file. MIDI data is a good solution because it can be easily transformed into music notation by standard programs such as

Logic Pro X or Finale. The authors have developed an open source C/UNIX-based program that automatically transforms a monophonic sound file into a playable MIDI file. Pitch (F0) detection is accomplished using a short-time autocorrelation algorithm. Successive F0's that correspond to the same MIDI note number are combined to form notes. The minimum duration of each note is determined by the autocorrelation window size, which in our case is set to 0.03 s. To achieve a more accurate notation result, the program employs duration and RMS amplitude thresholds to exclude spurious notes from the MIDI data.

11:30

2aMU10. A Spatially Distributed Vibrotactile Actuator Array (SDVAA) for music-to-vibrotactile sensory augmentation. Edgar J. Berdahl, Austin Franklin (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu), and Eric Sheffield (Music, Louisiana State Univ., Atlanta, GA)

The design of a Spatially Distributed Vibrotactile Actuator Array (SDVAA) is presented. It employs a multitude of vibrotactile actuators in order to communicate a larger amount of information than is possible using a single actuator. The SDVAA is currently being used for applications in music-to-vibrotactile sensory augmentation. While prior related projects have focused more on sensory substitution, this project aims only to add to a person's experience by augmenting a multimedia presentation with vibrotactile feedback. Because haptic perception is fundamentally different than auditory perception, it makes sense to rearrange the information transmitted to the haptic senses. For example, while auditory perception is limited to approximately the range 20 Hz to 20 kHz, tactile perception is limited primarily to the range 0 Hz to 800 Hz (if not higher). Accordingly, one approach being considered for converting auditory signals to vibrotactile signals is pitch shifting. Project results relating to music composed specifically for the SDVAA as well as general musical vibrotactile prototyping concepts will be presented.

TUESDAY MORNING, 14 MAY 2019 — SEGELL, 8:00 A.M. TO 11:15 A.M.

Session 2aNS

Noise, Architectural Acoustics, Structural Acoustics and Vibration, and ASA Committee on Standards: Structure-Borne Noise in Buildings and What We Can Do About It

Bonnie Schnitta, Cochair
SoundSense, LLC, 46 Newtown Lane, Suite One, East Hampton, NY 11937

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

Chair's Introduction—8:00

Invited Papers

8:05

2aNS1. Stucture-borne noise from a pool. Felicia Doggett (Metropolitan Acoust., LLC, 1628 JFK Blvd., Ste. 1902, Philadelphia, PA 19103, f.doggett@metro-acoustics.com)

A problem arose at a recently completed high-rise condominium building in Philadelphia. Tenants on many levels above a pool located on the second level of the building stressed that they could hear low-frequency sound when an early-morning swimmer was doing laps. Looking through literature, we found nothing to attest to the phenomenon that presented itself. To investigate, we used a 15 lb rubber medicine ball to simulate the cavitation caused by the swimmer, which also provided a repeatable source. Cavitation in the water created by the arm strokes of the swimmer was exciting the pool shell and transmitting through the structure ultimately converting to audible sound via the lightweight partitions in the living units. Our measurement system included time synchronized triaxial accelerometers to determine the transmission paths through the building, which identified two significant peaks at 35 Hz and 62 Hz. This case study details this unique problem, the identification of structural transmission, and the successful outcome after our recommendations were implemented.

8:25

2aNS2. Case study of acoustic disturbances in pencil towers due to building movement. Sean Harkin, Jennifer Scinto (Eng., SoundSense, LLC, P.O. Box 1360, Wainscott, NY 11975, sean@soundsense.com), and Bonnie Schnitta (Eng., SoundSense, LLC, East Hampton, NY)

Pencil structure high-rise residential towers are becoming increasingly prevalent in major city centers as the demand for housing increases in urban centers with limited available property for development. The narrow footprint of these structures, combined with the efforts to build subsequent structures higher than the last, lead to unique acoustic issues within these buildings, which impact the residents' quality of life and affect the value of the properties. During periods of high, and even medium wind speeds, these structures deflect more than traditional skyscraper buildings and the resulting moment force on the internal building assemblies creates disturbing acoustic events that have been measured to be significantly higher than ambient sound levels in select octave bands. This paper presents two case studies in one such pencil structure building in New York City, from diagnosis, to proposed and implemented innovative acoustic treatments, to client satisfaction.

8:45

2aNS3. Prevention of structural shorts in highly engineered noise and vibration isolation systems. Alexander C. Born (Getzner USA, Inc., 8720 Red Oak Blvd., Ste. 400, Charlotte, NC 28217, alexander.born@getzner.com) and Sean Harkin (SoundSense, LLC, Wainscott, NY)

Noise and vibration isolation in structures is becoming more and more common as the world we live in becomes noisier and noisier. The increase in noise coupled with people's desire for lower background noise and vibration criteria results in the need for higher performing isolation systems. In order to achieve these performance levels, the system needs to be properly designed and engineered. One of the most common culprits for a system not providing the expected performance calculated during the design process is a short or sound bridge . A short typically occurs during the installation of the system by rigidly connecting the structure or machine that is being isolated to an unisolated element. This causes noise and vibration from sources such as subway, transit, and mechanical equipment to transfer with ease into the structure intended for isolation, rendering the highly engineered system useless or highly degraded. This paper will examine a multitude of ways to eliminate the possibility of a short and elaborate on the importance of the installation process of isolated systems. It will also review how the process differs when isolating for vibration compared to noise and the importance of each.

9:05

2aNS4. Heavy-weight impact testing: Test repeatability for a modified kettlebell and comparison of impact source weights and drop heights. Michael Raley (Ecore Int., 715 Fountain Ave., Lancaster, PA 17601, mike.raley@ecoreintl.com)

Noise and vibration from heavy-weight impacts associated with fitness activities are a common source of complaints in hotels, multi-family residences, and corporate office spaces. *In situ* testing using various heavy/hard impact sources is commonly used to evaluate the effectiveness of mitigation measures to address noise and vibration complaints. Previous work (LoVerde *et al*., Internoise 2015) proposed the use of a 16-lb spherical shot for *in situ* testing. However, significantly heavier weights are common in many gyms. For instance, hotel chains commonly include free-weights with a range of 5 to 75 lb in their fitness rooms. It is not known if the relatively light 16-lb shot provides results that are indicative of impacts from significantly heavier weights. This presentation evaluates the use of heavier (50-lb and 100-lb) kettlebells and varying drop heights as alternates to the 16-lb shot. Additionally, this presentation evaluates test repeatability for the kettlebells which have been modified so that the impact surface is spherical in shape.

9:25

2aNS5. Too many Cooks in the kitchen?: A review of noise and vibration challenges in mixed-use buildings for the session: Structure-borne noise in buildings and what we can do about it. Sarah Taubitz (45dB Acoust., LLC, 45dB Acoust., LLC, PO Box 12275, Denver, CO 80212, st@45db.com)

Several projects with rooftop structural vibration and/or noise transmission through structures, and footfall impact issues will be shared, along with a review of recommendations and outcomes of the example projects. We plan to discuss noise/vibration challenges such as: (1) Design, or re-design, of Heating, Ventilation, and Air-conditioning (HVAC) and kitchen exhaust fan (KEF) on rooftops and within building envelopes. (2) Finding noise leaks in structure. (3) Is it the MRI machine's vibration, or the Variable Air Volume (VAV) box causing noise and vibration in the office upstairs? (4) Discussion of resonance of different floor/ceiling assemblies.

9:45–10:00 Break

10:00

2aNS6. Efficacy of high-performance ceiling systems. Wilson Byrick (Pliteq, 1370 Don Mills Rd., Unit 300, Toronto, ON M3B 3N7, Canada, wbyrick@pliteq.com) and Matthew V. Golden (Pliteq, North York, ON, Canada)

A current approach to try to solve a low frequency impact-induced structure-borne noise issue in lightweight woodframe buildings has been the application of a resilient ceiling suspended by one-inch deflection spring hangers. While this system has been lab tested it was only compared to an assembly with no isolation in which the gypsum board was directly attached to the bottom of the joists. The authors have repeated the testing of one-inch deflection ceiling hangers in the same lab but this time they compared it to the same isolators with the springs removed, standard resilient sound clips and other ceiling isolation systems. Relative performance differences of the isolation systems will be presented along with transmissibility, static stiffness, dynamic stiffness and damping measurements of each isolation element. A few theories as to why the systems behave the way they do will also be presented.

10:20

2aNS7. Measurements of rail vibration in a residential building behind a wave barrier trench. David W. Dong and John LoVerde (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com)

The authors have previously reported on a wave barrier trench that was constructed to reduce vibration transmission from railroad to a multifamily residential building [*J. Acoust. Soc. Am.* **139**, 2159]. The reduction in ground vibration levels due to the trench was measured in several site conditions [*J. Acoust. Soc. Am.* **140**, 3281]. This paper extends the study with additional measurements of vibration transmission into the building and structure-borne propagation within the building.

10:40

2aNS8. Developments in resonant, low frequency sound absorbing devices: trials, errors and successes. Jeffrey Madison (RPG Acoust. Systems, LLC, 99 South St., Passaic, NJ 07055, jmad008@hotmail.com)

Nowhere is it more prevalent in the acoustical treatment world where the predicted can deviate from the actual outcome than in resonant, low frequency sound absorber design and construction. Any worthwhile resonant sound absorber (RSA) is a system, multiple parts working together toward a common design goal. RSAs can be designed via mathematical prediction, Helmholtz much more accurately than membrane configurations. Regardless of the predictions, when the time comes to construct RSAs, each and every component that goes into and that must work with the other components in the system, whether in a factory setting or in field construction, will have a significant influence on the actual outcome. Prediction techniques swiftly identify resonant frequencies, for example, however can miss widely on bandwidth and efficiency due to variables introduced during construction. It is therefore important to use experimentation and measurement techniques to understand actual outcome possibilities. One example of this is utilizing a large format impedance tube that measures designs effectively down to 20 Hz. Examples of RSAs along with their development and testing will be presented.

Contributed Paper

11:00

2aNS9. Estimating the acoustic power of sources in semi-reverberant enclosures using generalized energy density. Travis N. Hoyt, Scott D. Sommerfeldt (Phys. and Astronomy, Brigham Young Univ., N306 ESC, Provo, UT 84602, travisnhoyt@gmail.com), and Jonathan Blotter (Mech. Eng., Brigham Young Univ., Provo, UT)

Sound power measurements of acoustic sources are typically performed in anechoic or reverberation chambers using acoustic pressure according to international standards. The anechoic chamber creates a free-field environment where the sound power is estimated from the squared pressure integrated over some enveloping surface. The reverberation chamber produces diffuse-field conditions, where sound power is proportional to the spatially averaged squared pressure. Since most acoustic sources exist in rooms that are neither anechoic nor entirely reverberant, it is desirable to estimate the sound power within these non-ideal, semi-reverberant spaces. In such environments, the direct and reverberant energies each contribute to the total measured field. If the kinetic and potential components of acoustic energy density are weighted appropriately, the spatial variation of the field can be significantly reduced compared to squared pressure. This generalized energy density allows an adaptation of the sound power formulation by Hopkins and Stryker to be used to make an efficient and accurate *in situ* sound power estimate of a noise source in a non-ideal acoustical environment. Since generalized energy density optimizes the spatial uniformity of the field, fewer measurement positions are needed compared to traditional standards. The experimental results and practical limitations of this method will be discussed.

Session 2aPA

Physical Acoustics and Noise: Nonlinear Acoustics for Non-Specialists I

Won-Suk Ohm, Cochair
Yonsei University, 50 Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea

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

Chair's Introduction—7:55

Invited Papers

8:00

2aPA1. Early history of nonlinear acoustics: Waveform distortion, disaster, and redemption. David T. Blackstock and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, dtb@austin.utexas.edu)

The first (although slightly incorrect) wave equations for finite-amplitude sound in lossless fluids were obtained independently by Euler in 1759 and Lagrange in 1760. Poisson (1808) provided the first major breakthrough with his exact solution for progressive waves of finite amplitude in a lossless gas. Although a far-reaching result, the progressive waveform distortion (and disastrous consequences) implied by his solution went unrecognized for 40 years. Challis (1848) showed that the Poisson solution is not single valued but did not understand why. Stokes (1848) provided the why. He saw that the Poisson waveform distorts as the wave travels, eventually threatening to become multivalued. He postulated that a discontinuity (shock) develops to avoid waveform overturning. He also saw that viscosity (not accounted for by Poisson) would prevent true discontinuities. Earnshaw (1860) and Riemann (1860) cleaned up plane waves in lossless gases. However, how to predict propagation after shocks form? Rankine (1870) and Hugoniot (1887, 1889) provided the first solutions for propagation when dissipation is included. These were the first steps toward redemption. The curtain rang down on this era of nonlinear acoustics with the excellent papers in 1910 by Rayleigh and Taylor on steady shocks in a thermoviscous fluid.

8:20

2aPA2. Overturning of nonlinear compressional and shear waveforms subject to power-law attenuation or relaxation. John M. Cormack and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

Model equations for plane nonlinear wave propagation that do not account for energy loss inevitably predict waveforms that are multivalued. Referred to as waveform overturning, this reveals that essential physics is not represented in the mathematical model. Energy loss from thermoviscous absorption keeps waveform steepening in check and prevents overturning for all source amplitudes, thus ensuring physical relevance of solutions. This is not the case for every energy loss mechanism, and waveform overturning may still occur above a critical source amplitude despite the presence of losses. The present work determines the critical source amplitude associated with two loss operators, one for power-law attenuation and the other for relaxation, and for both compressional waves (quadratic nonlinearity) and shear waves (cubic nonlinearity) [Cormack and Hamilton, *Wave Motion* **85**, 18 (2019)]. Reformulation of the model equations in intrinsic coordinates enables numerical solution of waveform evolution up to and beyond the distance at which waveform overturning occurs. It is found that attenuation and dispersion resulting from a power law with exponent less than unity, or the attenuation and dispersion due to relaxation, is insufficient to prevent waveform overturning at all amplitudes. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

8:40

2aPA3. Jet crackle: From production to propagation to perception. Kent L. Gee (Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

The sound of a crackling jet is unusual. It is an irregular, popping noise that gives a distinct impulsive quality to the broadband noise from military jet aircraft, rockets, and even volcanoes. What are its possible causes? Which factors influence its propagation—and it perception? This presentation reviews historical and recent research into a phenomenon that lies at the crossroads of fundamental physical and psychoacoustics.

9:00

2aPA4. Sonic booms and sonic thumps for non-specialists. Victor Sparrow (Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

One of the most straightforward applications of nonlinear acoustics is in the long range propagation of the shock waves created by supersonic aircraft. The sounds heard on the ground are either loud (sonic booms) or are quiet (sonic thumps) depending on the pressure versus time signature. This talk with give a brief overview of sonic booms and the role nonlinear acoustics plays in their propagation. The carpet of sounds heard on the ground from an example single flight will be shown. The role of nonlinear acoustics in the certification of future supersonic low-boom aircraft will be highlighted. Much more information about sonic boom is freely available from ntrs.nasa.-gov, and one great reference is "Sonic Boom" by Maglieri, *et al.* (NASA/SP-2014-622). [Work supported by the U.S. Federal Aviation Administration Office of Environment and Energy through ASCENT, the FAA Center of Excellence for Alternative Jet Fuels and the Environment, Project 41 through FAA Award Number 13-C-AJFE-PSU under the supervision of Sandy Liu. Any opinions, findings, conclusions or recommendations expressed in this material are those of the authors and do not necessarily reflect the views of the FAA.]

9:20

2aPA5. Nonlinear acoustics at Moscow State University. Oleg A. Sapozhnikov and Vera A. Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow 119991, Russian Federation, and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Washington, oa.sapozhnikov@physics.msu.ru)

The rapid development of nonlinear acoustics, which began in the second half of the 20th century, had several distinct centers in different countries, and one of them was M. V. Lomonosov Moscow State University (MSU) in Russia. From MSU came one of the basic equations in nonlinear acoustics—the Khokhlov-Zabolotskaya (KZ) equation, the 50th anniversary of which is celebrated this year [*Sov. Phys. Acoust.* **15**(1), 35–40 (1969)]. By the 1960–1970s, the work conducted by Academician Rem Khokhlov and his colleagues made MSU a leading international center for nonlinear acoustics and, even more influentially, for nonlinear optics. Nonlinear optics at that time underwent rapid growth related to the invention of lasers, and many phenomena in optics had analogs in acoustics. One of them was a parabolic approximation for wave beams, which led to formulation of the KZ equation after translation to acoustics with consideration of the details of nonlinear effects in weakly dispersive media. Along with the development of theoretical foundations, initial experiments on nonlinear acoustic phenomena were carried out at MSU. The current paper highlights these studies, as well as contemporary work and achievements on nonlinear acoustics at Moscow State University.

9:40

2aPA6. Audio spotlight: Sound from ultrasound. Joseph Pompei (Holosonics, 400 Pleasant St., Watertown, MA 02472, fjpompei@holosonics.com)

The Audio Spotlight® is the trade name of the first and only commercially successful ultrasonic loudspeaker, and delivers a laser-like beam of sound as tight as a beam of light. This allows audio to be delivered to one listener or a small group, without disturbing others nearby, or sound can be projected to create "true surround sound" without multiple loudspeakers. Originally developed at MIT, the technology is used widely in a variety of fields, such as museums, libraries, hospitals, and retail displays. This invited lecture will present an overview of the nonlinear acoustics as a basis for such a device, as well as some specific engineering challenges. Commercial applications will be discussed, and live demonstrations are included.

10:00–10:15 Break

Contributed Papers

10:15

2aPA7. Comparison of sonic boom propagation models with measurements above the Earth's turbulent boundary layer. Joel B. Lonzaga (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., B1208 MS 463, Hampton, VA 23681, joel.b.lonzaga@nasa.gov)

NASA conducted the Sonic Booms in Atmospheric Turbulence (SonicBAT) overflight campaign at Armstrong Flight Research Center in 2016 and Kennedy Space Center in 2017. The sonic booms were generated by F-18 aircraft flying at nearly constant altitudes and at nearly constant, supersonic speeds. Although the primary objective of the campaign was to investigate the effects of turbulence on the propagation of sonic booms from the aircraft, a subset of the SonicBAT datasets can be used to validate sonic boom propagation models. This dataset is obtained from the recording using a microphone mounted at a wingtip of a TG-14 glider flying above the Earth's turbulent boundary layer. With this dataset, effects due to turbulence and ground impedance, which are still poorly understood, are effectively eliminated. Consequently, comparison is made between measurements and models which only account for the nonlinear and absorption effects. Such models are obtained using NASA's PCBoom sonic boom propagation code, which has recently been updated to incorporate the full wind effects and improve the code's accuracy and efficiency. Additionally, near-field signatures of the aircraft using PCBoom's near-field approximant are also compared with those generated by The Boeing Company using computational fluid dynamics.

10:30

2aPA8. Finite difference time domain investigation of interior sound fields generated by parametric acoustic arrays. Anpeksh A. Saksena and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., Columbus, OH 43210, saksena.13@osu.edu)

Parametric acoustic arrays enable tight beams of audible sound through the nonlinear collimation and mixing of ultrasonic acoustic waves. Previous investigations have considered parametric array characterization when operated in free field environments and in the presence of rigid planar reflectors, so that the opportunity to leverage parametric arrays in more realistic environments with diffusive scatterers and absorptive surfaces remains unexplored. To fill the knowledge gap, this research establishes a finite difference time domain (FDTD) model of a parametric array in an interior environment with a combination of reflective, absorptive, and diffusive bounding surfaces. The modeling framework builds upon existing FDTD models of nonlinear ultrasound wave propagation used in the life sciences and medicine. Here, the studies seek to elucidate how the sonic sound field

is manipulated when the reflective surfaces that receive the nonlinear ultrasonic waves are neither planar nor perfectly reflective. The relative ability to take advantage of such interaction between incident ultrasound and reflected energies, when compared to a perfectly rigid plane for reflection, is examined in detail. All together, this work establishes and harnesses an FDTD modeling framework to examine the use of parametric acoustic arrays in conventional interior environments having reflective, absorptive, and diffusive surfaces.

10:45

2aPA9. Nonlinear focusing of high amplitude sound using time-reversal. Brian D. Patchett, Brian E. Anderson, Matthew L. Willardson, and Michael Denison (Phys. & Astronomy, Brigham Young Univ., N203, Eyring Sci. Ctr., Provo, UT 84606, brian.d.patchett@gmail.com)

Time reversal (TR) is a signal processing technique that may be used to intentionally generate high amplitude focusing of sound. The use of time reversal in room acoustics has been studied by others, but the application to generating high amplitude focusing has not previously been explored. The purpose of this study is to generate high amplitude sound waves in order to mimic a virtual spherical source with enough intensity to observe nonlinear wave propagation. Experiments have been carried out in a reverb chamber with eight compression horn drivers. Using these drivers, the impulse response is calculated, reversed in time, and modified using the clipping technique. When these signals are broadcast from the sources, a focus is generated at the receiver location with peak levels reaching 198 dB (ref 20 μPa). As the waves superpose at the focus, the amplitudes observed do not scale linearly when the experiment is repeated at increasing amplification levels. The compression peaks are higher in amplitude than expected, and the rarefaction troughs are lower in amplitude (less negative) than expected, when compared to linear scaling. Additionally, the diverging waves from the focus resemble the propagation of a single spreading shockwave.

11:00

2aPA10. Comparison of two ultrasonic backscatter coefficient methods under nonlinear distortion. Andres Coila and Michael Oelze (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Mathews, Urbana, IL 61801, acoila@illinois.edu)

The backscatter coefficient (BSC) is a fundamental property of tissues and can be parameterized for tissue characterization. The BSC requires a reference signal, which is estimated through either the planar reflector method (PRM) or the reference phantom method (RPM). In both methods, linear acoustic propagation is assumed. In this work, the BSC estimation methods are evaluated when nonlinear distortion is present. RF data were acquired from two physical phantoms, labeled A and B, with a 5 MHz single-element transducer using low power (1 excitation level) and high power (6 increasing excitation levels) excitation signals. Phantom A contained glass beads with diameters ranging from 75 to 90 μm and phantom B had glass beads with diameters ranging from 9 to 43 μm. The BSCs estimated using the low power setting and high power settings were compared for the both the PRM and RPM through a root mean square error (RMSE). Estimates of the effective scatterer diameter (ESD) were obtained using each method for each high power setting and compared to the low power setting. The RMSE increased as the power setting increased with much higher RMSEs using the PRM compared to RPM, i.e., a maximum of 7.3 times larger for phantom A and 8.6 times larger for phantom B. Estimates of ESD matched the ranges of glass beads sizes for both phantoms except when using the PRM at higher power settings. These findings suggest that the RPM is more robust to nonlinear distortion effects compare to the PRM.

11:15

2aPA11. Enhancing dynamic positioning performance inside mid-air acoustic levitator. Tatsuki Fushimi (Mech. Eng., Univ. of Bristol, Queen's Bldg., Bristol BS8 1TR, United Kingdom, t.fushimi@bristol.ac.uk), Asier Marzo (UpnaLab, Universidad Pública de Navarra, Bristol, United Kingdom), Thomas L. Hill, and Bruce W. Drinkwater (Mech. Eng., Univ. of Bristol, Bristol, United Kingdom)

Acoustic levitators are devices which generate converging acoustic radiation forces and thus can trap objects in mid-air. Acoustic levitators have found applications in the fields of chemistry and biology as a non-contact transportation method. The trapping of objects can be achieved using a phased-array in which the phase of the signal sent to each transducer is varied to generate a trap that stably holds the particle at the target three-dimensional position. The transducer phases can be changed over time to translate the acoustic field in space, thereby transporting the trapped particle. Here, we describe an open-loop spatial calibration scheme which increases the positioning accuracy of a particle in an acoustic levitator. The effectiveness and the performance of the spatial calibration was determined using a single-axis standing wave levitator with 60 ultrasonic transducers (40 kHz), and a levitated particle (EPS particle of radius 0.7 mm). Our calibration method is shown to significantly improve the positioning accuracy of the particle inside the acoustic levitator and reduced RMS error down to 0.11 and 0.03 mm in x and z axes, respectively. Although the calibration approach only considers the static response, the trajectory when the particle moves at a relatively high velocity ($\approx$1 cm/s) was also improved. Increasing the precision and velocity of a moving particle will enhance the capabilities and reliability of acoustic levitators and open up possibilities for novel applications.

11:30

2aPA12. Born approximation of acoustic radiation force and torque on soft objects of arbitrary shape. Thomas S. Jerome, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029, tsjerome@utexas.edu)

When the density and compressibility of an object are similar to the corresponding properties of the surrounding fluid, and the dominant contribution to the acoustic radiation force depends on gradients of the energy densities as occurs in plane standing waves, the Born approximation may be used to calculate both the radiation force and torque. The approximation consists of integrating the monopole and dipole contributions to the force throughout the volume of the object, and thus it is applicable to objects with arbitrary shapes and material property distributions. Here, an axisymmetric object in a plane standing wave is considered, resulting in one-dimensional integral expressions for the radiation force and torque. The integrals are evaluated analytically for homogeneous spheres and cylinders. The accuracy of the approximation is assessed via comparison with full solutions for spheres and prolate spheroids based on eigenfunction expansions for the corresponding scattering problem. Different densities and compressibilities of the object and the surrounding medium, as well as different sizes, shapes, and orientations of the object relative to the standing wave field, are considered. Limitations of the approximation and potential extensions to more complex wave fields are discussed. [T.S.J. supported by the ARL:UT McKinney Fellowship in Acoustics.]

11:45

2aPA13. Simulation of the second harmonic ultrasound field in heterogeneous soft tissue using a mixed domain method. Juanjuan Gu and Yun Jing (Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., EB III, campus box 7910, Raleigh, NC 27695, jgu4@ncsu.edu)

A mixed-domain method dubbed frequency-specific mixed domain method is introduced for the simulation of the second harmonic ultrasound field in weakly heterogeneous media. The governing equation for the second harmonics is derived based on the quasilinear theory. The speed of sound, nonlinear coefficient, and attenuation coefficient are all spatially varying functions in the equation. The fundamental frequency pressure field is first solved by the frequency-specific mixed domain method, and it is subsequently used as the source term for the second harmonics equation. This equation can be again solved by the frequency-specific mixed domain method to rapidly obtain the second harmonic pressure field. Five two-dimensional cases, including one with a realistic human tissue map, are studied to systematically verify the proposed method. Results from the previously developed transient mixed domain method are used as the benchmark solutions. Comparisons show that the two methods give similar results for all cases. More importantly, the frequency-specific mixed domain method has a crucial advantage over the transient mixed domain method in that it can be two orders of magnitude faster.

Session 2aPPa

Psychological and Physiological Acoustics and Education in Acoustics: Acoustics Outreach to Budding Scientists: Planting Seeds for Future Clinical Physiological Collaborations

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

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

Chair's Introduction—8:00

Invited Papers

8:05

2aPPa1. Effects of noise-induced hearing loss on speech-in-noise envelope coding: Inferences from single-unit and non-invasive measures in animals. Satyabrata Parida (Weldon School of Biomedical Eng., Purdue Univ., 715 Clinic Dr., Lyles-Porter Hall, West Lafayette, IN 47907, spsatyabrat@gmail.com) and Michael G. Heinz (Speech, Lang., and Hearing Sci. & Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN)

Speech-intelligibility models (SIM) can be used for systematic fitting of hearing-aids and cochlear-implants, potentially improving clinical outcomes in noisy environments. Existing SIMs are suitable for predicting performance of normal-hearing subjects, but not for hearing-impaired subjects due to our limited understanding of the effects of cochlear hearing impairment on speech-in-noise coding. In this work, we collected auditory-nerve (AN) single-unit responses and envelope following responses (EFR) in normal- and hearing-impaired chinchillas to speech, spectrally-matched stationary-noise, and noisy-speech. Our data show increased correlation between AN-fiber response envelopes of noisy-speech and noise-alone for hearing-impaired fibers in speech-relevant modulation-frequency bands, suggesting a greater degree of distraction from inherent envelope fluctuations following cochlear hearing loss. This novel finding is significant given the emphasis recent SIMs [e.g., Jørgensen and Dau, *JASA* (2011)] have placed on the importance of inherent noise-envelope fluctuations in addition to speech-coding fidelity in predicting noisy-speech perception. Preliminary data also show enhanced fundamental-periodicity coding at the expense of place-specific formant coding, and a degradation of burst envelopes of high-frequency fricatives for the hearing-impaired group. EFRs show evidence for degraded tonotopic coding, as observed in single-unit responses [e.g., Henry *et al.*, *J. Neurosci.* (2016)]. [Work supported by Action on Hearing Loss (UK).]

8:25

2aPPa2. Passive music listening: A modulation of resting-state functional connectivity to better dissociate tinnitus. Somayeh Shahsavarani, Yihsin Tai (Speech and Hearing Sci., Inst. for Adv. Sci. and Technol., Univ. of Illinois at Urbana-Champaign, 2514 Beckman 405 N Mathews Ave., Urbana, IL 61801, bahar@illinois.edu), Sara Schmidt, Rafay Khan (Neurosci. Program, Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Fatima Husain (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Previous fMRI studies have shown tinnitus-related changes in resting-state functional connectivity (rs-fc) and have the potential for serving as biomarkers of tinnitus. In this study, we further investigated the effect of tinnitus on three intrinsic neural networks—the auditory network, the dorsal attention network (DAN), and the default mode network (DMN) while the participants were at rest or under a passive music listening condition. Our results indicated that listening to music altered the auditory network in both groups, with increased connectivity between inferior frontal gyrus and auditory areas in controls. This alternation was observed only in the patients with mild tinnitus handicap and was absent in the patients with severe tinnitus handicap. The effect of music on the DAN hinged upon hearing sensitivity: decreased connectivity between the lateral occipital cortex and the DAN, and increased connectivity between the precuneus and the DAN was observed in controls and patients with normal hearing, compared to those with hearing loss. Furthermore, passive music listening modulated the coherency of the DMN based on tinnitus status and/or hearing sensitivity. Our findings highlight the efficacy of rs-fc in dissociating the relatively heterogeneous tinnitus population and its subgroups from controls, using rest and listening to music.

8:45

2aPPa3. Evaluating human neural envelope coding as the basis of speech intelligibility in noise. Vibha Viswanathan (Weldon School of Biomedical Eng., Purdue Univ., 715 Clinic Dr., Lyles-Porter Hall, West Lafayette, IN 47907, viswanav@purdue.edu), Hari M. Bharadwaj (Speech, Lang., & Hearing Sci. and Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN), Barbara Shinn-Cunningham (Neurosci. Inst., Ctr. for the Neural Basis of Cognition, Biomedical Eng., Psych., and Elec. & Comput. Eng., Carnegie Mellon Univ., Pittsburgh, PA), and Michael G. Heinz (Speech, Lang., & Hearing Sci. and Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN)

Models of speech intelligibility that accurately reflect human listening performance across a broad range of background-noise conditions are clinically important (e.g., for deriving hearing-aid prescriptions, and optimizing cochlear-implant signal processing). A leading hypothesis in the field is that internal representations of envelope information ultimately determine intelligibility. However, this hypothesis has not been tested neurophysiologically. Here, we address this gap by combining human electroencephalography (EEG) with simultaneous perceptual intelligibility measurements. First, we derive a neural envelope-coding metric (*ENVneural*) from EEG responses to speech in multiple levels of stationary noise, and identify a mapping between the neural metric and corresponding speech intelligibility. Then, using the same mapping, we use only EEG measurements to test whether *ENVneural* is predictive of speech intelligibility in novel background-noise conditions and in the presence of linear and non-linear distortions. Preliminary results suggest that neural envelope coding can predict speech intelligibility to varying degrees for different realistic listening conditions. These results inform modeling approaches based on neural coding of envelopes, and may lead to the future development of physiological assays for characterizing individual differences in speech-in-noise perceptual abilities.

9:05

2aPPa4. Effect of noise reduction on cortical speech processing in hearing aid users. Subong Kim, Inyong Choi, and Yu-Hsiang Wu (Univ. of Iowa, 250 Hawkins Dr., Iowa City, IA 52242, subong-kim@uiowa.edu)

Noise reduction (NR) has been widely used in hearing aids (HAs) to increase ease and comfort of listening and to reduce listening effort. However, NR attenuates noise at the potential cost of distorting speech cues. This makes it challenging for audiologists to select the best configuration for NR during HA fitting process. The long-term goal of our research is to optimize HA fitting by characterizing the neural mechanisms underlying the effect of NR. The purpose of the present study is to examine the effect of NR on cortical dynamics during speech-in-noise tasks in HA users using electroencephalography. Our recent study with normal-hearing listeners has shown that speech recognition in low-level noise engaged greater early activity (~300 ms after word onset) in left supramarginal gyrus and weaker late activity (~700 ms) in left inferior frontal gyrus, than in high-level noise. Based on these findings, we hypothesized that, for a given patient, the optimal NR configuration would be the one that can recruit this "low-level noise" pattern of neural activity. Initial results from the electroencephalographic source space analysis will be presented, and underlying cortical mechanisms of speech processing in HA users will be discussed.

9:25

2aPPa5. Displacement of the stapes differs across species—Implications for studies of auditory function. John Peacock, Mohamed Alhussaini, Nate Greene, and Daniel J. Tollin (Univ. of Colorado, 12800 E 19th Ave., Aurora, CO 80045, john.2.peacock@ucdenver.edu)

Sound is transferred to the cochlea via the middle ear. The anatomy and physiology of the middle ear varies significantly across species, and these differences impact both the stimulation provided to the inner ear, and the suitability of different animal models for use in various types of research. Studies of auditory trauma from blast, for example, require generation of intracochlear pressures with sufficiently high intensities to cause damage. In some species, e.g., mice and rats, it may not be possible to generate sufficiently high pressures through air conducted sound alone, whereas in humans sufficiently high pressures can readily be generated through air conduction. We hypothesize that this is due to limits on the displacement of the stapes by the stapedial annular ligament, which thereby constrains the energy transferred to the cochlea through the middle ear. To test this hypothesis, we made measurements of the motion of the middle ear bones in response to tones of varying intensities and frequencies in several different species commonly used in laboratory research. Our results reveal peak stapes displacements from ~150 um in humans to 10-20 um in mice and rats. We will discuss the implications of these findings for basic studies of auditory function.

9:45

2aPPa6. Simultaneous measurement of electrocochleography and ear canal pressure in normal-hearing adults. Jessica Chen, Michael Simpson, and Skyler G. Jennings (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1217 BEH S, Salt Lake City, UT 84112, jessica.chen@utah.edu)

The auditory system is regulated by various adaptive mechanisms that modify sound as it passes upstream to the cortex. Studying these adaptive mechanisms, such as the middle ear muscle (MEM) reflex and the medial olivocochlear (MOC) reflex, reveals how the normal auditory system adjusts to challenging listening tasks and how abnormal reflexes may contribute to perceptual difficulties experienced by patients with sensorineural hearing loss. Studies in laboratory animals reveal that MOC and MEM reflexes result in neural antimasking at the output of the auditory nerve. Antimasking effects at the level of the middle ear and at the level of the cochlear hair cells should carry over to effects observed at the output of the auditory nerve. Antimasking due to MEM and MOC reflexes has been difficult to study in humans since measurement approaches are limited to non-invasive techniques with only a single output variable (e.g., ear canal pressure). In these studies, we assess MEM and MOC function by simultaneous measurements of ear canal pressure and electrocochleography, including the cochlear microphonic potential and compound action potential. Simultaneous measurements are expected to reveal the antimasking effects of MEM and MOC reflexes to middle ear, hair cell, and auditory nerve responses.

Session 2aPPb

Psychological and Physiological Acoustics: Topics in Physiological and Psychoacoustics (Poster Session)

Gregory M. Ellis, Chair
Communication Sciences and Disorders, Northwestern University, Department of Communication Sciences and Disorders, Evanston, IL 60201

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

Contributed Papers

2aPPb1. Effects of age, modulation rate, and modulation depth on sentence recognition in speech-modulated noise. Daniel Fogerty, Rachel E. Miller (Commun. Sci. and Disord., Univ. of South Carolina, 1224 Sumter St., Columbia, SC 29208, fogerty@sc.edu), Jayne B. Ahlstrom, and Judy R. Dubno (Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Speech is often heard in amplitude modulated backgrounds when speech is glimpsed during momentary masker dips. Increased masker modulation depth provides more improvement in the signal-to-noise ratio of glimpsed speech, but also improves noise modulation detection for potentially greater modulation masking. Mechanisms of modulation masking and glimpsing may also depend on the overlap of the modulation spectra of the speech and noise. This study employed modulation filtering and noise amplitude compression to investigate the combined effects of noise modulation depth and modulation rate on speech recognition. Younger normal-hearing (YNH), older normal-hearing (ONH), and older hearing-impaired (OHI) adults listened to sentences in noise that were spectrally shaped to control for individual hearing thresholds. A second YNH group listened to sentences with the same spectral shaping as OHI listeners. Sentence recognition generally improved with greater noise modulation depth, especially at higher modulation rates. Results suggest that speech recognition for all groups is maximized when speech modulations <8 Hz are preserved, when noise modulation occurs at rates higher than this range, and with greater noise modulation depth. OHI listeners benefit similarly from these conditions, but their poorer overall performance may be due to reduced sensation levels. [Work supported by NIH/NIDCD.]

2aPPb2. Psychophysical and anatomical evidence for hidden hearing loss in laboratory mice. Kali Burke (Psych., Univ. at Buffalo, SUNY, 246 Park Hall, Buffalo, NY 14260, kaliburk@buffalo.edu), Laurel A. Screven (Otolaryngol. – Head and Neck Surgery, Johns Hopkins Univ. School of Medicine, Baltimore, MD), Anastasiya Kobrina (Psych., Univ. at Buffalo, SUNY, Amherst, NY), Katrina M. Schrode, Amanda Lauer (Otolaryngol. – Head and Neck Surgery, Johns Hopkins Univ. School of Medicine, Baltimore, MD), and Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, Buffalo, NY)

Exposure to high intensity sound can lead to temporary or permanent threshold shifts. Noise exposures that do not cause long-term hearing deficits, however, can induce extensive afferent ribbon synapse loss, while hair cells and spiral ganglion neurons remain mostly intact (Kujawa and Liberman, 2009). This loss of synapses despite normal hearing thresholds is referred to as hidden hearing loss (HHL). We examined the development of HHL in laboratory mice using operant conditioning with positive reinforcement. After exposing mice to 8–16 kHz narrowband noise at 100 dB SPL for 2 h, hearing thresholds temporarily shifted for both pure tone and ultrasonic vocalization stimuli; however, post-exposure thresholds and threshold shifts varied by sex and age. Immunohistochemistry and transmission electron microscopy were conducted to quantify peripheral damage and central synaptic reorganization once behavioral testing was complete. Brains were collected, sectioned, and labeled against VGLUT1 or GAD65 and labeling was quantified in the ventral cochlear nucleus. Cochleas were also collected, dissected, and labeled for myosin6 to label hair cells and either CTBP2 or SV2 to identify afferent or efferent terminals, respectively. Our findings show that mice are able to behaviorally recover hearing following non-traumatic noise exposure despite changes in peripheral and central auditory structures.

2aPPb3. The effect of broadband elicitor duration on transient-evoked otoacoustic emissions and a psychoacoustic measure of gain reduction. William Salloom (Speech, Lang., and Hearing Sci., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, wsalloom@purdue.edu), Hari M. Bharadwaj, and Elizabeth A. Strickland (Speech, Lang., and Hearing Sci. and Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN)

Physiological and psychoacoustic studies of the medial olivocochlear reflex (MOCR) in humans have often relied on long elicitors (> 100 ms). This is largely due to previous research using otoacoustic emissions (OAEs) that found MOCR time constants in the 100s of milliseconds when elicited by broadband noise. However, Roverud and Strickland (2014), using a psychoacoustic measure of gain reduction, found differential effects of duration for on- and off-frequency tonal elicitors. For the on-frequency elicitor, thresholds increased with increasing on-frequency duration up to about 50 ms, and then plateaued. In contrast, thresholds with off-frequency elicitors continued to increase with elicitor duration. These results are consistent with cochlear gain reduction, possibly by the MOCR, in which the on-frequency elicitor is affected by gain reduction at the signal frequency place, but the off-frequency elicitor is not. The effect of the duration of broadband noise elicitors on similar psychoacoustic tasks is currently unknown. Additionally, the relationship between gain reduction measured psychoacoustically and using OAEs as a function of elicitor duration are unknown. This study will measure the effects of ipsilateral broadband noise elicitor duration on transient-evoked OAEs and psychoacoustic gain reduction estimated from a forward-masking paradigm. [Work supported by NIH R01 DC008327 (EAS) and NIH R01DC015989 (HMB).]

2aPPb4. Physiological studies of binaural tone-in-noise detection in the inferior colliculus of awake rabbit. Langchen Fan, Kenneth S. Henry, and Laurel H. Carney (Univ. of Rochester, MC 5-7616 601 Elmwood Ave., Rochester, NY 14642, langchen.fan@gmail.com)

With identical noise maskers presented to both ears, human listeners have lower tone detection thresholds when the tone is presented out-of-phase between the ears (N_0S_π) rather than in-phase (N_0S_0). The threshold difference is called the binaural masking level difference (BMLD). Human listeners have BMLDs up to at least 4 kHz, but previous studies to understand neural mechanisms have only used low-frequency tones. Here, we recorded single-neuron responses from the inferior colliculus in awake rabbit to a wide range of overall noise levels and tone frequencies near the neuron's characteristic frequency. Maskers were 1/3-octave gaussian noise centered at the tone frequency. Neural thresholds at each noise level were estimated from average-rate responses at signal-to-noise ratios of -12 to +8 dB based on receiver-operating-characteristic analysis. Neural thresholds were similar across noise levels and were pooled together for further analysis. BMLDs of all recorded neurons with measurable thresholds for both N_0S_0 and N_0S_π ranged up to +17 dB (i.e., a substantial masking release for the N_0S_π condition). Across the population of neurons tuned to different frequencies, the largest BMLDs at each frequency decreased with increasing frequency, consistent with human psychophysical studies, but with a more gradual slope.

2aPPb5. Influence of viscous boundary layer on wave propagation on isolated tectorial membranes. Haiqi Wen, Charlsie Lemons, Elisa Boatti, and Julien Meaud (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta 30339, GA, hwen@gatech.edu)

The tectorial membrane is an extracellular matrix located in the cochlea. The tectorial membrane is often hypothesized to play an important role in hearing mechanics. Measurements of wave propagation on isolated tectorial membranes have been used in the literature to characterize the intrinsic mechanical properties of tectorial membranes in the auditory frequency range. While most previous studies have made an implicit assumption regarding the width of the tectorial membranes in order to find the properties of the TM using a simple model, we have recently used a more accurate model that takes into account the finite width of the TM and its anisotropy. However, experiments are conducted in an artificial endolymph bath, which we neglected in our previous analysis. In this work, we study the influence of the viscous boundary layer due to the fluid on wave propagation on isolated tectorial membranes. The boundary layer adds damping and mass to the tectorial membrane, which we model using a commercial finite element software. The influence of the boundary layer on the space constant and wave speed of the longitudinally propagating radial motion, and on the spatial patterns of the longitudinal motion, are analyzed.

2aPPb6. Implementing an efferent signal into an auditory pathway model for tone-in-noise detection. Afagh Farhadi (Elec. and Comput. Eng., Univ. of Rochester, 1900 Wind Willow Way Apt. 11, Rochester, NY 14624, afarhadi@ur.rochester.edu) and Laurel H. Carney (Biomedical Eng. & Neurosci., Univ. of Rochester, Rochester, NY)

Simultaneous tone-in-noise detection has been studied extensively, but typically without consideration of the medial olivocochlear (MOC) efferents. We are testing hypotheses for masked detection using a central auditory model with a signal from midbrain to MOC. Masked tones are encoded in the rate profile of band-enhanced (BE) inferior colliculus (IC) neurons, which are excited by a range of modulation frequencies. Peripheral responses to noise are characterized by large fluctuations, an effective stimulus for BE IC neurons. In contrast, peripheral channels tuned near the tone have smaller fluctuations: addition of the tone flattens the signal envelope and also pushes the inner hair cell (IHC) transduction nonlinearity further into saturation. Excitatory projections to MOC from noise-driven BE IC cells would decrease cochlear gain, reducing IHC saturation, and resulting in larger fluctuations and IC rates. In contrast, tone-plus-noise-driven channels would reduce MOC excitation, resulting in relatively higher cochlear gain, more saturation, and ultimately lower IC rates. Thus, the descending signal from IC BE cells to MOC is hypothesized to enhance contrast in the IC rate profile. Because efferents have slow dynamics, timing is an important factor. Therefore, we focus on model sensitivity to masked tones of different durations for comparison to psychophysical trends.

2aPPb7. Detection and intensity discrimination of a sinusoid with and without masker uncertainty. Christopher Conroy, Christine Mason, and Gerald Kidd (Dept. of Speech, Lang. & Hearing Sci. and Hearing Res. Ctr., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, cwconroy@bu.edu)

The effect of stimulus-based uncertainty on the shape of the masking function at circa-threshold pedestal levels was investigated. Of particular interest was how uncertainty regarding properties of the masker affects the magnitude of "negative masking" [Raab *et al.*, *J. Acoust. Soc. Am.* **35**, 1053 (1963)] that is obtained under different stimulus configurations. Intensity discrimination thresholds for gated, 100-ms, 1000-Hz sinusoids were measured at three pedestal levels: -9, 0, and 9 dB re: absolute threshold. A two-interval, two-alternative forced-choice procedure was used. Under the reference condition, thresholds were measured in quiet. In comparison conditions, thresholds were measured in the presence of one of two masker types: (1) a notched-noise masker or (2) a random-frequency, multicomponent masker. In the multicomponent masker condition, uncertainty was imposed by roving the frequency components that comprised the masker from interval-to-interval over a block of trials. The data under each condition were fit with psychometric functions and slope estimates were obtained. Results are discussed with respect to the factors that govern, and possible mechanisms underlying, negative masking. Specifically, two possible explanations for the "dip" in the masking function are examined: nonlinear transduction [Hanna *et al.*, *J. Acoust. Soc. Am.* **80**, 1335 (1986)] and observer uncertainty.

2aPPb8. Sensitivity to AM incoherence is affected by center frequency and modulation rate. Kelly L. Whiteford and Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, whit1945@umn.edu)

Fine-grained sensitivity to frequency modulation (FM) at slow rates and low-frequency carriers is thought to be due to auditory-nerve phase locking (time code). Alternatively, a unitary code for FM at all rates and carrier frequencies could be based on cochlear conversion of FM to amplitude modulation (AM) (place code). One weakness of the place-coding theory is it cannot readily explain rate- and carrier-dependent trends in FM sensitivity. This study asked whether FM trends could potentially be explained by sensitivity to two AM envelopes that are out of phase (incoherent AM) at separate cochlear locations, thereby simulating the effects of FM. AM discrimination was assessed for two-component complexes centered at low (500 and 1500 Hz) and high (7000 Hz) frequencies, spaced 2/3 or 4/3 octaves apart, and modulated at slow (2 Hz) and fast (20 Hz) rates. Coherent and incoherent two-component AM detection was assessed for the same conditions. Preliminary results show that sensitivity to AM incoherence is best at low center frequencies and slow rates, consistent with trends traditionally found in FM detection that have been attributed to time coding. Findings suggest time coding may not be necessary to explain trends in FM sensitivity. [Work supported by NIH Grant R01DC005216.]

2aPPb9. Discrimination of rippled spectra: Contribution of excitation-pattern and temporal-processing mechanisms. Olga Milekhina, Dmitry Nechaev, and Alexander Supin (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru)

Rippled-spectrum signals are used for measurements of signal resolution in hearing-impaired listeners and cochlear-implant users. Two mechanisms may be responsible for rippled-spectrum resolution. The excitation-pattern mechanism determines the ripple density resolution (ripples/oct). The temporal-processing mechanism determines the ripple frequency spacing limit (kHz). Contributions of the mechanisms can be assessed by comparison of resolutions of band-limited rippled spectra with different center frequencies, because the ratio of rippe spacing to ripple density is frequency-proportional. Ripple-density resolutions of half-octave rippled spectra were measured at center frequencies from 0.5 to 4 kHz. The

measurements were performed either by discrimination between rippled-spectrum test and reference signals differing by ripple phases or by discrimination between a rippled-spectrum test and non-rippled reference signal. For discrimination between rippled-spectrum test and reference signals, resolution specified in ripples/oct little depended on center frequency, as predicted by the excitation-pattern model. For discrimination between rippled test and non-rippled reference signals, the resolution specified in ripple frequency spacing little depended on center frequency, as predicted by the temporal-processing model. It was concluded that contributions of the excitation-pattern and temporal-processing mechanism depend on the discrimination task.

2aPPb10. Rapid measurement of frequency depending spectro-temporal modulation detection thresholds. Stefan Klockgether, Julia Rehmann, and R. Peter Derleth (R&D, Sonova AG, Laubisrütistrasse 28, Stäfa, Zürich 8712, Switzerland, stefan.klockgether@sonova.com)

The modulation detection threshold of spectro-temporal modulations (STM) is highly correlated with the speech reception threshold (SRT). A common interpretation of this correlation is that detection capability of concurrent spectral and temporal modulations is a measure for the frequency resolution power of the auditory system. The STM-detection threshold can be measured for different frequency regions by applying the modulation only to particular frequency bands. One could make an estimation of the frequency resolution power at those particular frequency regions. This could give a hint on a beginning or a hidden hearing loss which is not showing up clearly in an audiogram. This study shows different methods to rapidly measure STM-detection thresholds frequency dependent and compares the results with results gained with an established adaptive staircase method. The rapid methods are evaluated with regard to their practicability and the trade-off between reduced duration and reduced accuracy.

2aPPb11. The duplex theory re-revisited: Spectral weighting of localization cues in tones and noises. G. Christopher Stecker, Monica L. Folkerts, and Julie M. Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu)

It is widely suspected that sound localization is accomplished primarily through the use of low-frequency interaural-time-difference (ITD) cues and only secondarily via high-frequency ITD and level-difference (ILD) cues [Rayleigh, *Philos. Mag.* **13**, 214–232 (1907); Wightman and Kistler, *JASA* **91**, 1648-1661 (1992); Macpherson and Middlebrooks *JASA* **111**, 2219–2236 (2002)]. Contemporary studies of cross-frequency interactions in spatial hearing have provided support for this view but have not directly identified the frequencies involved nor quantified the relative weighting of binaural cues weighting across components of a single complex. This study adapted the temporal-weighting approach of Stecker and Hafter [*JASA* **112**, 1046–1057 (2002)] to measure spectral weighting functions (SWF) for (a) free-field and reverberant sound localization and (b) lateralization based on ITD and/or ILD. Across a wide range of conditions, SWFs featured enhanced weights for components in the vicinity of 400-800 Hz, supporting a narrow "dominance region" for localization and lateralization of complex sounds [Bilsen and Raatgever, *Acoustica* **28**, 131–132 (1973)], and a precipitous drop in weights between 800 and 1400 Hz [Brughera *et al.*, *JASA* **133**, 2839–2855 (2013)]. [Work supported by NIH R01-DC016643.]

2aPPb12. Consequences of interaural decorrelation of speech temporal fine structure. Lucas Baltzell, Virginia Best, and Jayaganesh Swaminathan (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, lbaltzel@bu.edu)

The ability to understand a target speech signal against a background of interfering speech signals is typically improved when the interfering signals are spatially separated (spatial release from masking; SRM). Swaminathan *et al.* (2016) found a significant reduction in SRM when the temporal fine structure (TFS) across the left and right ears was decorrelated, suggesting that binaural TFS provides important cues that support SRM. One interpretation is that degrading binaural TFS prevents the extraction of reliable interaural time differences (ITDs), which in turn leads to a reduction in SRM. We tested this hypothesis by systematically decorrelating the binaural TFS, and measuring the effects on both ITD discrimination for speech stimuli and SRM in normal hearing listeners. We show that decorrelation leads to a systematic increase in ITD discrimination thresholds, as well as a systematic decrease in SRM. This supports the idea that binaural TFS is needed to access ITD information in speech, which is in turn required for SRM. Additionally, the relationship between these two tasks provides a framework for determining the extent to which reduced SRM in listeners with hearing loss can be attributed to reduced sensitivity to binaural TFS.

2aPPb13. Decoding listener's attention :Can it be improved with behavioural measure of selective attention? Moïra-Phoebé Huet (Univ. of Lyon, 25 Ave. Jean Capelle O, LVA, Villeurbanne 69100, France, moira-phoebe.huet@insa-lyon.fr), Christophe Micheyl (Starkey, Lyon, France), Etienne Gaudrain (Univ. of Lyon, Groningen, The Netherlands), and Etienne Parizet (Univ. of Lyon, Villeurbanne, France)

During the past decade, there has been growing interest in the neural correlates of selective attention to speech. In these studies, listeners were instructed to focus their attention on one of two concurrent speech streams. However, in everyday-life situations, a listener's attention can switch rapidly between different voices. Thus, we have developed a behavioural protocol to infer the dynamics of auditory attention over time. After listening to two simultaneous stories—a target and an interferer—the participants have to find, among a set of words, those present in the target story. The participant's responses are then used to estimate, retrospectively, when their attention was directed toward the target, or toward the interferer. Neural data, recorded with EEG, and behavioural measures are combined to extract the brain's temporal response function in response to these stimuli. Moreover, to promote attention switches between the two voices, the interferers were uttered by the same talker as the target stories, but the voice parameters were manipulated to parametrically control the similarity of the two voices. We will discuss the results in terms of attentional selection and voice confusion, and suggest possible applications of this dynamic behavioral test of selective auditory attention.

2aPPb14. Ability of normal hearing listeners to recognize vowels and musical instruments under spectrally-degraded conditions. Ryan Anderson, Alyxandria Sundheimer, and William Shofner (Speech and Hearing Sci., Indiana Univ., 200 S Jordan, Bloomington, IN 47401, anderyan@indiana.edu)

Music perception requires greater spectral resolution than speech perception [Shannon, *Int. Rev. Neurobiol.* **70**, 121 (2005)]. However, conclusions from these metadata are problematic given that they aggregate results from several different studies using diverse methodologies and paradigms. In particular, methodologies generally tapped into different perceptual dimensions, namely word recognition for speech and melody recognition for music. The present study aims to develop a paradigm to compare speech and music recognition based on similar perceptual dimensions, namely timbre. Stimuli consisted of naturally-spoken vowels and notes played on musical instruments as well as 32-, 8-, and 4- channel noise-vocoded versions. Listeners discriminated either instruments from vowels or vowels from instruments using a go/no-go task. The discrimination paradigm offered insight as to how available spectral information influenced perception between stimulus sound categories. Reaction times and accuracy were measured and organized as a function of signal degradation level to consider potential differences in how normal hearing listeners utilize spectral fine structure information when distinguishing vowels and musical instruments. In general, listeners' ability to discriminate vowels from instruments under degraded conditions was similar to their ability to discriminate instruments from vowels. The results suggest that the perception of vowels and musical instruments rely on similar mechanisms.

2aPPb15. Pure-tone thresholds and the binaural masking level difference. Daniel E. Shub, Joshua G. W. Bernstein, Lina R. R. Kubli, Douglas S. Brungart (National Military Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., 4954 N. Palmer Rd., Bethesda, MD 20889, joshua.g.bernstein.civ@mail.mil), and Ken W. Grant (National Military Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Silver Spring, MD)

Although several studies have examined the relationship between high-frequency pure-tone thresholds and the 500-Hz binaural masking level difference (BMLD), the results have not always been consistent. In this study, a retrospective analysis was conducted on an existing dataset from over 3000 military service members that included both pure-tone thresholds measured as part of their annual hearing conservation testing and a 33-trial clinical test of the BMLD. There was only a slight dependence of the 500-Hz BMLD on high-frequency pure-tone thresholds. For listeners with elevated pure-tone thresholds, this dependence was in good agreement with the findings of Jerger *et al*. [*Arch. Otolaryngol*. **110**, 290–296 (1984)] and slightly larger than that reported by Wilson and Weakley [*J. Am. Acad. Audiol*. **16**, 367–382 (2005)]. For listeners with near-normal hearing, the dependence of the 500-Hz BMLD on the 4-kHz pure-tone threshold was substantially less than that reported by Bernstein and Trahiotis [*J. Acoust. Soc. Am*. **140**, 3540–3548 (2016)]. A possible explanation might be the degree of training offered to the subjects and procedural differences between clinical and laboratory techniques. [The views expressed in this abstract are those of the authors and do not reflect the official policy of the Department of Army/Navy/Air Force, Department of Defense, or U.S. Government.]

2aPPb16. The fine-grained statistical structure of speech is congruent with nonlinear peripheral auditory processing. François Deloche (CAMS, PSL Univ., 54 Blvd. Raspail, Paris 75006, France, francois.deloche@ehess.fr)

Efficient coding of sensory signals takes advantage of statistical regularities in sensory data. Cochlear filter in mammals are known to reflect the overall statistical structure of speech, in line with the hypothesis that low-level sensory processing provides efficient codes for information contained in natural stimuli. Recently, some efforts have been made to describe this correspondence in more detail. The study of the statistical structure of speech over different acoustic classes demonstrates that frequency selectivity should not be fixed to achieve maximum efficiency. On the other hand, cochlear signal processing is nonlinear as frequency selectivity decreases with sound intensity level. Both effects are greater in the high frequencies. In the present study, these two facts are shown to be consistent in the case of a parametric method based on Gabor dictionaries (Gaussian-modulated sinusoids) and in a simplified setting. A model with fewer constrains is also introduced for future experiments to validate this hypothesis in a more general context.

2aPPb17. Harmonicity aids detection of speech and other sounds in noise. Malinda J. McPherson (Div. of Medical Sci., Harvard Univ., MIT Bldg. 46-4078, 43 Vassar St., 46-4078, Cambridge, MA 02139, malindamcpherson@g.harvard.edu) and Josh McDermott (Dept. of Brain and Cognit. Sci., Massachusetts Inst. of Technol., Cambridge, MA)

Acoustic grouping cues, such as the tendency of frequencies to be harmonic, are used to segregate multiple sounds, as when listening to one of several concurrent speakers. However, we often must listen to sound sources in noise. Here we investigate the role of traditional acoustic grouping cues in detecting sound sources in noise. We measured detection thresholds for several types of sounds embedded in noise: speech, musical instruments, and synthetic complex tones. In each case the sounds were resynthesized with harmonic and inharmonic carrier frequencies to test the importance of harmonic frequency structure for hearing in noise. We found that harmonic signals were consistently easier to detect in noise than otherwise similar inharmonic signals. This harmonic advantage persisted even when the phases of harmonic components were randomized, such that the advantage is unlikely to reflect differences in the depth of modulation (from beating). Near threshold, harmonic signals were readily audible in conditions where inharmonic signals could not be heard. These results suggest that harmonicity is critical for detecting real-world signals in noise, demonstrating its relevance to another important aspect of auditory scene analysis.

2aPPb18. Does lateral position explain release from informational masking arising from interaural time and level differences? Richard L. Freyman, Casey D. Milkey (Dept. of Commun. Disord., Univ. of Massachusetts, Amherst, MA 01003, rlf@comdis.umass.edu), Emily Buss (Otolaryngology/Head and Neck Surgery, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Patrick Zurek (Sensimetrics Corp., Malden, MA)

Spatial release from masking (SRM) is known to arise from the interaural time and level differences (ITDs and ILDs) produced by spatially separated signal and masker sources. In some situations, SRM might be augmented considerably by the differing spatial perceptions created by the signal and masker, which could help alleviate confusions between them. Using headphone presentation, the present study investigated this hypothesized perceptual component of SRM. Maskers were binaural click trains of approximately 2-s overall duration with inter-click intervals (ICIs) varying randomly between 50 and 150 ms. The ITD or ILD of the masker clicks was manipulated across listening blocks. The listeners' task was to identify the number of signal clicks, identical to the masker clicks except for their ITD/ILD, inserted during masker inter-click intervals. Preliminary testing with a fixed masker ICI demonstrated that the signal clicks were easily audible, strongly suggesting that difficulty during the main identification task was due to confusions from the randomized ICI. The study explored whether performance counting signal clicks in the randomized ICI conditions could be explained by the differences in intracranial spatial perception created by combinations of ITD and ILD cues. [Work supported by NIDCD R01 01625.]

2aPPb19. Revisiting superoptimal perceptual integration for pitch at high frequencies. Hedwig E. Gockel and Robert P. Carlyon (MRC Cognition and Brain Sci. Unit, Univ. of Cambridge, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom, hedwig.gockel@mrc-cbu.cam.ac.uk)

In a frequency discrimination task, Lau *et al*. [*J. Neurosci.* (2017)] reported superoptimal integration of information from individual components in a very high frequency region, where phase locking to the temporal fine structure is presumably absent, when the components were combined into a harmonic complex. We tried to replicate this finding using stimuli identical to those of Lau *et al*., with some additional conditions. Using an adaptive two-alternative forced-choice procedure, we measured fundamental frequency difference limens (F0DLs) for complex tones containing harmonics 6–10 with F0s of 280 and 1400 Hz, and frequency difference limens (FDLs) for each harmonic of the complex presented alone. Stimulus duration was 210 and 1000 ms. All tones had random phases, a ± 3 dB level rove, and were presented in a continuous threshold-equalizing noise that was either diotic or dichotic. Observed DLs were lower overall than in the study of Lau *et al*. As in their study, for the low F0, observed F0DLs were worse than predicted assuming optimal combination of frequency information from the individual harmonics and assuming that performance is limited by peripheral noise. However, for the high F0, observed and predicted F0DLs did not differ significantly, in contrast to the finding of Lau *et al*.

2aPPb20. Age-related differences in modulation detection interference and interference release. Yuan He and Jennifer Lentz (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, heyuan@indiana.edu)

Age-related declines have been observed in auditory tasks related to temporal processing, but less work has been conducted in middle-aged listeners and in auditory sound segregation tasks. This experiment addresses these issues by evaluating modulation detection interference (MDI) and release from MDI using streaming in young, middle-aged and older listeners. Using a standard MDI paradigm, we measured amplitude modulation detection thresholds of a high-frequency tone in different conditions: in quiet, in the presence of a low-impact unmodulated interferer, and in the presence of a high-impact modulated interferer. Then, we measured streaming-based release from MDI using four modulated precursors designed to perceptually capture the interferer and therefore reduce the amount of interference. Preliminary data suggest that middle-aged normal-hearing listeners experience similar MDI to young listeners and also receive a large release from interference when precursors are present. Additional data will be presented from older listeners, who experience slightly greater MDI than younger listeners.

2aPPb21. Blast exposure in the military and its effects on sensory and cognitive auditory processing. Scott Bressler (Biomedical Eng., Boston Univ., 610 Commonwealth Ave., Rm. 923C, Boston, MA 02215, bressler@bu.edu), Kimberly Jenkins, Jennifer Myers, Kenneth Grant (Audiol., Walter Reed National Military Medical Ctr., Bethesda, MD), and Barbara Shinn-Cunningham (Carnegie Mellon Univ., Pittsburgh, PA)

Blast-induced traumatic brain injury (TBI) and hearing loss are the two most common types of injuries sustained by military personnel while serving in the U.S. Global War on Terrorism. Recently several VA audiology clinics have reported active duty service members complaining of having problems communicating in noisy listening environments despite having normal to near-normal pure tone thresholds. In addition to standard clinical measures, we used electroencephalography (EEG) to determine whether damage to suprathreshold responding auditory nerve fibers in the sensory periphery and/or trauma to cortical regions associated with attention and working memory were responsible for the reported listening complications. In separate auditory and visual selective attention tasks, behavioral and neural measures suggest no evidence of long term neurotrauma affecting normal cognitive function. We found while absolute measures of auditory brainstem encoding varied greatly in all study subjects, comparisons of how the envelope following response (EFR) changes with modulation depth hint at differences between blast and non-blast exposed service members. These findings are consistent with audiometric threshold and distortion product otoacoustic emission data that show subtle differences between groups within clinically defined normal limits. Taken together these results suggest subclinical differences in audiometric measures might explain differences in suprathreshold listening.

2aPPb22. A harmonic-cancellation-based model to predict speech intelligibility against a harmonic masker. Luna Prud'homme, Mathieu Lavandier (Laboratoire Génie Civil et Bâtiment, Univ Lyon, ENTPE, rue Maurice Audin, Vaulx-en-Velin 69518, France, luna.prudhomme@entpe.fr), and Virginia Best (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA)

This study aimed at predicting speech intelligibility in the presence of harmonic maskers. Contrary to a noise signal, these maskers have a harmonic structure that allows for a segregation of the competing sounds based on a difference of their fundamental frequency (F0). This F0 segregation could be due to spectral glimpsing or harmonic cancellation. It is unclear what the relative contributions of these two mechanisms are. In this work, we have modified the model of Collin and Lavandier [*J. Acoust. Soc. Am.* **134**, 1146–1159 (2013)] in order to take into account both mechanisms. Different implementations of the model were compared and applied to two data sets: SRTs measured for monotonous harmonic maskers that varied in their F0s and degree of harmonicity; SRTs measured for monotonous and intonated harmonic maskers that varied in their mean F0, F0 contour, temporal envelope and spatial position. Comparison of data and model predictions will allow us to establish how to implement harmonic cancellation in our model framework, to evaluate to what extent the model can predict F0 segregation for harmonic maskers, and to determine the relative roles of spectral glimpsing and harmonic cancellation suggested by the model.

2aPPb23. Backward recognition masking of similar and dissimilar environmental sounds. Kelli Clark, Allison Holtz, Nirmal Kumar Srinivasan (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., 8000 York Rd., Towson, MD 21252, kclark44@students.towson.edu), Jeremy Gaston, and Brandon Perelman (Human Res. and Eng. Directorate, U.S. Army Res. Lab., Adelphi, MD)

Backward Recognition Masking (BRM) of sound occurs when two sounds are presented close in time to one another and the second sound hinders the recognition of the first sound. Previous studies on BRM used either white noise or sine tones. Here, we present backward recognition thresholds using environmental sounds. 34 young normal hearing individuals were presented with 6 combinations of similar or dissimilar environmental sounds from the categories of household appliances, automobiles, and power tools. The sounds were selected based on their multi-dimensional scaling distances (Rosen *et al.*, 2017). The sounds were presented at 20 dB SL re: 3 frequency PTA and the inter stimulus interval was fixed at 22.2 ms. A two-down one-up adaptive procedure was used to identify the target duration at which the listeners could identify that the target is different from the masker. Initial data analyses indicated that similar sounds needed significantly longer presentation durations to be identified as different compared to dissimilar sounds, indicating more backward masking for similar sounds compared to dissimilar sounds. These results potentially highlight how critical safety information in real-world environments may be missed in complex listening scenarios.

2aPPb24. The effects of modulator shape and methods for expressing modulation depth on spectral modulation detection thresholds. Sittiprapa Isarangura, Katherine Palandrani (Univ. of South Florida, 4202 E. Fowler Ave. PCD1017, Tampa, FL 33620, sisarangura@mail.usf.edu), Trevor Stavropoulos, Aaron Seitz (Univ. of California, Riverside, Riverside, CA), Eric C. Hoover (Dept. of Commun. Sci. and Disord., Univ. of Maryland, College Park, MD), Frederick J. Gallun (VA RR&D National Ctr. for Rehabilitative Auditory Res., Portland, OR), and David A. Eddins (Univ. of South Florida, Tampa, FL)

The detection of sinusoidal modulation is commonly used for assessing the auditory perception of temporal, spectral, and spectro-temporal acoustic features. For temporal (amplitude) modulation, the sinusoidal modulator usually is expressed on a linear amplitude scale. For spectral modulation, the sinusoidal modulator has been specified on a linear amplitude scale, consistent with temporal modulation, or on a logarithmic amplitude scale, with the notion of approximating a sinusoidal excitation pattern. The definition of modulation depth depends on the measurement points (i.e., midpoint to peak or peak to peak) and order of operations when expressing depth in dB. Such differences can make it difficult to compare results for similar tasks among investigations. Here we quantify differences among methods and provide a complete matrix for translating among methods. Spectral modulation detection was measured for 9 normal-hearing listeners in ten conditions (linear vs. logarithmic shape at 0.5, 1, 2, 4, and 8 cycles/octave). Peak-to-peak values of the modulation envelope were equalized, thus only modulation waveform shape differed. Noise carriers had a passband from 400 to 3200 Hz. Thresholds revealed statistically significant effects of both spectral shape and spectral modulation frequency. Several methods for expressing threshold modulation depth were compared to highlight differences among the methods.

2aPPb25. Pre-trial and post-trial cueing of masker location in a localization-in-noise task. Brian Simpson (Air Force Res. Lab., 2610 Seventh St., Area B, Bldg. 441, Wright-Patterson AFB, OH 45433, brian.simpson.4@us.af.mil), Robert H. Gilkey, Michelle Wang (Dept. of Psych., Wright State Univ., Dayton, OH), Nathaniel Spencer, and Eric R. Thompson (Air Force Res. Lab., Wright-Patterson AFB, OH)

Localization accuracy for a target presented in a simultaneous masker, whose location varies randomly from trial to trial, improves when a preview

of the masker location is provided (by playing a sound from that location) prior to the target + masker interval (i.e., a pre-trial cue) [B. Simpson, Ph.D. dissertation (Wright State University, 2011)]. One explanation is that knowing the masker location allows a listener to establish a "spatial attention filter" at the masker location. The present study compares the effect of such a pre-trial cue to the case in which the cue comes *after* the target + masker interval (post-trial cueing). That is, the cue is presented either 500 ms prior to the onset of a 60-ms, 100-Hz click-train target embedded in a 60-ms broadband masker, or 500 ms subsequent to the offset of the target + masker stimulus. The data indicate that both cue types lead to similar improvements in performance over the no-cue condition, with the greatest improvement from cueing (~6 dB) seen for localization in the left/right dimension. While these data are roughly consistent with previous results, they cannot be explained by a simple spatial filter hypothesis.

2aPPb26. The effect of simulated Doppler frequency changes on detectability of complex tones. Lawrence L. Feth, Evelyn M. Hoglund, Omkar Dixit (Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, feth.1@osu.edu), and Matthew Davis (CDO Technologies, Columbus, OH)

The detectability of frequency changes designed to simulate the Doppler effect for moving sound sources has been well documented in the literature (Chowning, 1977; Ericson, 2001; Kaczmarek, 2005, Getzmann, 2008, and Pörschmann & Störig, 2009). The current study was designed to determine whether the presence of Doppler frequency shifts have a measurable effect on the detectability of complex sounds. Detection thresholds for single sinusoids with frequency changes ranging from the frequency DL to a full octave are compared with those for complex tones with selected components frequency modulated to simulate Doppler effects. A one-interval, adaptive tracking paradigm using the SIAM procedure (Kaernbach, 1991) was used to determine three points on the psychometric functions so that detection thresholds and slopes could be compared. For single component signals, the direction of frequency change and the span of the frequency sweep appear to have an effect on detectability. Results for complex tones require further explanation. [Work supported by a contract from CDO Technologies, Dayton, OH.]

2aPPb27. Masking of short tones in noise: Evidence for envelope-based, rather than energy-based detection. Skyler G. Jennings and Jessica Chen (Commun. Sci. and Disord., Univ. of Utah, 390 S. 1530 E., Salt Lake City, UT 84112, skyler.jennings@hsc.utah.edu)

A short tone added to a longer simultaneous masker may be detected by observing an increase in overall acoustic energy. This detection scheme predicts probe thresholds to be independent of the masker's temporal envelope. A recent study [Jennings *et al.*, *J. Assoc. Res. Oto.* **19**, 717–727 (2018)] revealed that probe thresholds were 10 dB higher for maskers with flattened compared to fluctuating envelopes, suggesting an envelope-based detection strategy. This study test the hypothesis that probe thresholds are proportional to envelope power by measuring detection thresholds for a 4-kHz, 6-ms probe in one of three temporal positions within a 400-ms masker in normal-hearing listeners. The narrow-band, low-fluctuating noise masker was preceded by flattened or fluctuating noise precursors. The precursor's offset was delayed from the masker's onset by -2, 0, 10, 25, 100, or 250 ms. Probe thresholds were positively correlated with envelope power (R-squared = 0.75), consistent with masking from precursor envelope fluctuations and from fluctuations introduced by precursor and masker ramps. These findings suggest that future modeling efforts for short probes presented in longer maskers should consider a decision variable based on envelope fluctuations.

2aPPb28. Surveying the sounds used in the Journal of the Acoustical Society of America (1950–2017). Michael Schutz and Jessica Gillard (School of the Arts, McMaster Univ., 1280 Main St. West, Hamilton, ON L8S4M2, Canada, schutz@mcmaster.ca)

The earliest auditory psychophysical experiments involved naturalistic sounds such as hammers striking plates. The subsequent development and ubiquity of desktop computing gave researchers the ability to more precisely control stimulus parameters such as frequency, amplitude, and duration (Neuhoff, 2004). However much of our everyday listening is for events rather than easily manipulated properties (Gaver, 1993), and the world lacks the kinds of constrained sounds often used in auditory research (Phillips *et al.*, 2002). Although simplistic auditory stimuli hold benefits with respect to control, their disproportionate use poses problems for generalizing outcomes from key experiments. To provide insight into the sounds used in auditory perception research, we surveyed a representative sample of auditory stimuli from 217 psychophysical experiments published in JASA between 1950 and 2017. Our survey documents a disproportionate focus on simplistic sounds, with less than 4% of psychophysical experiments using stimuli exhibiting the dynamic temporal structures characteristic of natural auditory events. We will discuss the implications of these findings in the content of ongoing areas of inquiry of broad relevance to the auditory perception community.

2aPPb29. Auditory detection and sound source elevation. M. Torben Pastore and William Yost (College of Health Solutions, Arizona State Univ., 954 East Lobster Trap Ln., Tempe, AZ 85283, m.torben.pastore@gmail.com)

The spectral profile of a given sound stimulus varies with the spatial location according to the head-related transfer functions (HRTF) at both ears. For narrowband and tonal stimuli at frequencies above approximately 4 kHz, perceived sound source elevation has been shown to depend mainly on stimulus frequency instead of sound source location (e.g., Blauert, Morimoto), and this result has in turn been shown to correlate with spectral features of the HRTF (e.g., Middlebrooks). However, even though the HRTF is important to perceived elevation, listeners do not generally report a perceptual awareness of the spectral filtering that occurs as a result of the HRTF, suggesting that some form of spectral equalization may occur at some level beyond peripheral auditory processing. Therefore, depending on the level at which detection processing occurs, we might expect different detection thresholds for high frequency stimuli presented from different elevations, or, conversely, we might expect them to be the same. We will report data, collected in a soundfield, concerning the interaction, if any, of sound source elevation and auditory masked detection of high frequency stimuli.

2aPPb30. Spatial external noise and detection. M. Torben Pastore (College of Health Solutions, Arizona State Univ., 4 Irving Pl., Troy, New York 12180, m.torben.pastore@gmail.com) and William Yost (College of Health Solutions, Arizona State Univ., Tempe, AZ)

In vision, a change in target position can cause an otherwise stationary target to perceptually "pop out," thereby increasing its detectability. Previous work in this lab and others has shown that there is no corollary effect for auditory detection. The explanation most often given for this "null result" is that, while vision is encoded spatiotopically, audition is encoded tonotopically, and therefore changes in sound source position should not be expected to impact auditory detection in the way that target movement affects visual detection. This argument assumes that detection is largely a peripheral task. However, Green, Watson, and others have shown that auditory detection is affected by listener uncertainty about the frequency of presented target tones—this "external noise," in terms of frequency, suggests that detection involves processing at higher levels beyond the periphery (e.g., attention). It therefore stands to reason that auditory target stimuli may be less detectable when sound source location is randomized—a spatial manifestation of external noise. We will present data testing this hypothesis.

2aPPb31. Talker head orientation discrimination using only auditory cues. David L. Frazier (Speech and Hearing Sci., Univ. of Illinois, 1506 Pine Trace Ct, University Park, IL 60484, supermanfrazier@gmail.com) and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois, Champaign, IL)

Although many studies focus on human ability to localize a sound source, less is known about human ability to determine the physical orientation of a given sound source. In our study, we assessed listeners' ability to detect changes in talker head orientation. Participants with normal hearing were asked to detect head orientation changes (relative to 0 deg, *i.e.*, directly facing the listener) for two male and two female talkers. We found listeners

are sensitive to changes of approximately 40 deg in talker head orientation using only auditory cues. This is less sensitive than what humans have displayed with only visual cues. These findings indicate that auditory cues are available for head orientation discrimination, which may be of greater utility when visual cues are unavailable.

2aPPb32. Measured localization performance at low stimulus levels with an adaptive tracking task. Nathaniel Spencer (Airforce Res. Lab., 2610 7th St., Area B, Bldg. 441, Wright-Patterson Airforce Base, OH, spencernj80@gmail.com), Eric R. Thompson, and Brian Simpson (Airforce Res. Lab., Wright-Patterson AFB, OH)

Sabin *et al.* (2005) measured localization accuracy for 250-ms broadband noises that varied in location along a spherical surface, and varied in stimulus level, from 0 to 60 dB above the detection threshold. They found that, for lower stimulus levels, responses tended to be biased towards lower elevations and to the front, and were more accurate at higher stimulus levels. Whereas Sabin *et al.* (2005) randomized level and location between trials, the current study fixed location and increased level until a correct response was given (or the 80 dB SPL limit was reached). Our initial level was 12.5 dB SPL, and our step size 2.5 dB SPL. Responses tended to be biased towards lower vertical elevations and to the front at low stimulus levels, and be more accurate at higher levels, consistent with Sabin *et al*. The percentage of front/back errors was generally greater for locations in the rear hemifield relative to those in the frontal hemifield, and for locations at higher elevations relative to locations at lower elevations. Audibility analysis, performed using a KEMAR manikin, showed that localization errors tended to decrease the most when audible bandwidth increased at the acoustically better ear.

2aPPb33. Updating of spatial selective auditory attention under-compensates for listener head movement. Ewan A. Macpherson, Miyoung Jeon, and Serena Ransom (National Ctr. for Audiol., Western Univ., 1201 Western Rd., Elborn College 2262, London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

Listeners can use spatial selective auditory attention (SSAA) to focus on one talker in a complex acoustic scene. In our dynamic SSAA task, listeners oscillate their heads ~±40 deg at ~0.5 Hz while five different simultaneous sequences of four spoken digits are presented from loudspeakers at 0 deg azimuth (the target) and ±22.5 deg and ±45 deg azimuth (four distractors); listeners report the target sequence heard. We have observed [ASA, Minneapolis 2018] that listeners are more likely to misreport distractors that are centrally located in head-centered coordinates at the moment of presentation, and that under static conditions, performance declines with increasing target eccentricity—suggesting either that listeners cannot rapidly update the focus of their SSAA to compensate head motion (“lag”) or that they have difficulty directing SSAA eccentrically (“low gain”). To differentiate these alternatives, spatio-temporal maps of SSAA, conditioned on head orientation and direction of motion during each digit, were derived by computing the percentage of reported digits corresponding to those emitted by each loudspeaker. The spatial pattern of errors depended primarily on head orientation and not on direction of motion, suggesting that in this task SSAA tends to remain centrally focused in head-centered coordinates (low gain) rather than lagging dynamic head position.

2aPPb34. Development of duration discrimination during adolescence. Jennifer Gay, Merri Rosen (Dept. of Anatomy and Neurobiology, Northeast Ohio Medical Univ., Rootstown, OH), and Julia J. Huyck (Speech Pathol. and Audiol., Kent State Univ., 1325 Theatre Dr., Kent, OH 44242, jhuyck@kent.edu)

Temporal processing, which is important for comprehending speech, matures over an extended developmental period. Here we investigated duration discrimination during adolescence. Listeners aged 8–19 years (four age groups) heard three broadband noises on each trial, and indicated which, if any, of the noises was different in length (longer or shorter). The broadband noises were 15, 30, 50, 100, or 200 ms in duration. During each of two sessions on consecutive days, each combination of stimulus durations (e.g., 15 vs. 100 ms) were presented a total of twenty times in pseudo-randomized order, including identical comparisons (e.g., 100 vs. 100 ms) to enable calculation of false alarm rates. All age groups showed a higher (better) sensitivity (d’) for comparisons whose durations were more different from one another (e.g., 15 vs. 200 ms). Nevertheless, duration discrimination abilities continued to develop up to age ~14 years: 8- to 10-year-olds did not differ from 11- to 13-year-olds, but 11- to 13-year-olds had lower sensitivity than 14- to 17-year-olds and young adults, who themselves did not differ from one another. There was no learning between sessions and the interactions were not significant. Thus, even simple duration discrimination abilities may continue to develop into adolescence.

Session 2aSA

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

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

Alexey S. Titovich, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd., West Bethesda, MD 20817

Invited Papers

8:00

2aSA1. Low frequency sound absorption by compliant Helmholtz resonators. Shichao Cui and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., Columbus, OH 43210, cui.408@osu.edu)

A traditional Helmholtz resonator is a rigid-walled cavity and an open neck, which result in one acoustic resonance. In this study, a soft-walled cavity with an open neck, a compliant Helmholtz resonator, is investigated for broadband and large absorption of sound at low frequencies. By applying the soft materials, the dynamics of the compliant walls and the material impedance are coupled to the acoustic pressure filed in the cavity. Thus the impedance of the compliant resonator system as seen by an incident acoustic wave is determined by the interaction of the wall dynamics, material properties, and acoustic domain. Compared to a traditional Helmholtz resonator with the same geometric dimensions, the compliant Helmholtz resonator exhibits multiple resonances with large absorption of sound. In addition, strategic design of the structural composition and the material selection leads to resonant behavior at frequencies lower than the conventional Helmholtz resonance. Through exploring an analytical model developed to characterize the resonator, the threshold on material and structural compliance to yield traditional Helmholtz behavior is also illuminated.

8:20

2aSA2. Evaluation of metamaterial unit cell analysis techniques. Amanda Hanford, Benjamin Beck, Aaron J. Stearns, and Andrew S. Wixom (Appl. Res. Lab, Pennsylvania State Univ., P.O. Box 30 - MS 3230D, State College, PA 16804, ald227@psu.edu)

The field of acoustic metamaterials has produced novel materials in a wide variety of applications. An important step in designing a metamaterial is unit cell analysis with subwavelength geometry. There are several techniques used for unit cell analysis when designing acoustic properties of interest for metamaterial applications. Such techniques include, but are not limited to, band diagrams, effective material properties, or half-space homogenization. This talk discusses the challenges and tradeoffs between analysis techniques and types of structures that lend to one method or another. Unit cell analysis methods will be evaluated to perform trade space exploration, including validation and parametric studies for metamaterial design.

8:40

2aSA3. Passive Non-Reciprocity in Asymmetrical, Hierarchical, Nonlinear Metamaterials. Michael J. Leamy, Amir Darabi, Lezheng Fang, Matthew Fronk (Georgia Tech, 771 Ferst Dr. NW, Atlanta, GA 30332-0405, michael.leamy@me.gatech.edu), and Alex Vakakis (Univ. of Illinois Urbana-Champaign, Urbana, IL)

Reciprocity is a property of linear, time-invariant systems whereby the energy transmission from a source to a receiver is unchanged after exchanging the positions of the source and receiver. Non-reciprocity, on the other hand, violates this property and can be introduced to systems if time-reversal symmetry and/or parity symmetry is lost, or by introducing nonlinearity. While many studies have induced non-reciprocity by active means, considerably less attention has been given to acoustical structures that passively break reciprocity. In this talk, we will discuss passive, strongly nonlinear periodic structures which exhibit giant reciprocity breaking under impulsive and/or harmonic excitation. Numerical means are employed to generate dispersion curves, as a function of wave energy, which differ for left-to-right from right-to-left propagation. These dispersion curves become reciprocal at the limiting cases of low and high energy, which can be shown analytically. In between, varying degrees of non-reciprocity can be achieved, and in some regimes, giant reciprocity breaking is achieved with very little harmonic distortion. This suggests many possibilities for passive, non-reciprocal devices that operate with near-linear behavior.

Contributed Papers

9:00

2aSA4. Contrast parameter characterizes Bragg frequency gap in periodic flexural beams. Thomas Gallot (Instituto de Física, Facultad de Ciencias, Universidad de la República, Igua 4225, Montevideo 11400, Uruguay, tgallot@fisica.edu.uy), Adrien Pelat, and Francois Gautier (Université du Mans, Laboratoire d'Acoustique de l'Université du Maine, Le Mans, France)

Periodic structures exhibit frequency bands where destructive interferences prohibit wave propagation. Such behavior can be useful to mitigate vibrations, in particular when structure lightening is important. Band gap properties can be deduced from Bloch's theory approach, Plane Wave Expansion or Multiple Scattering methods. These methods require a full description of the unit cell geometry and its mechanical properties. In this work we focus on an ideal unit cell geometry to highlight the role of a contrast parameter in the band gap opening process. We consider flexural waves in beams and demonstrate analytically that the contrast parameter fully controls the first Bragg band gap. Numerical simulation and experiments on a beam demonstrator proves that the gap bandwidth is independent of the section geometry: only the flexural rigidity is involved. We propose a semi-analytic model for the central frequency gap that depends on the mass distribution. The established algebraic expressions for the band gap bandwidth and central frequency successfully matches the results in the practical case and can be used to design flexural wave cut-band filters. Finally, symmetry considerations explains the experimental observation of a second band gap, also suitable for vibration mitigation, due to coupling of flexural and compressional waves.

9:15

2aSA5. Theoretical study of a metasurface-based sound absorber. Jun Ji (Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., EBIII, Raleigh, NC 27695-7910, jji5@ncsu.edu), Ni Sui, Xiang Yan (Mech. and Aerosp. Eng., North Carolina State Univ., Providence, RI), Fuh-Gwo Yuan, and Yun Jing (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

A metasurface-based sound absorber panel is proposed to achieve high sound absorption with a deep sub-wavelength thickness and a relatively broad bandwidth. The panel composed of different periodically-arranged Helmholtz resonators has been designed by a radiation impedance model which takes the mutual radiation impedance into account. Numerical simulations based on COMSOL Multiphysics show that the radiation impedance model yields more accurate results compared with the traditional equivalent surface impedance method. Broad bandwidth of sound absorption is achieved under proper coupling by carefully tuning the geometry and distance between Helmholtz resonators.

9:30

2aSA6. A Lagrangian formulation of metamaterial homogenization. Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil)

Many metamaterials may be well described as multi-scale systems where the large scale is represented by the wavelength and the small scales are represented by inhomogeneity lengths. The effective material properties of a metamaterial of this sort may be derived by considering the averaged behavior of material elements that are considered "infinitesimal" relative to the wavelength but is on the order of the inhomogeneity lengths. The averaging of the small-scale features enables a straightforward process to describe the metamaterial mechanics in terms of a Lagrangian formalism, and the macroscopic constitutive equations may be obtained by deriving the Euler-Lagrange equations for the macroscopic field quantities. This energetic approach to homogenization may be used to estimate both linear and nonlinear material properties. This talk describes this homogenization method for one-dimensional systems and compares the resulting effective material properties with other homogenization methods.

9:45–10:00 Break

10:00

2aSA7. Transverse to longitudinal mode conversion via nonlinear inertial effects in a locally resonant metamaterial plate. Samuel P. Wallen and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, walles@uw.edu)

Acoustic and elastic metamaterials have potential for tailoring wave propagation in ways beyond the capabilities of conventional materials and have been shown to exhibit a myriad of rich dynamical effects, such as negative effective properties, cloaking, extreme damping, and non-reciprocity. However, most of the existing literature has been focused on linear dynamics in which these behaviors are restricted to narrow frequency bands. More recently, nonlinearity has been investigated as a method of increasing this bandwidth, as well as a means to broaden the palette of available dynamics. Past studies on nonlinear acoustic and elastic metamaterials have demonstrated effects not easily accessible in the linear regime, including harmonic generation, mode hopping and conversion, tunable band gaps, chaos, and intrinsic localized modes. In this work, we present a model for a nonlinear elastic plate metamaterial that exhibits mode conversion between transverse and longitudinal waves. The mode conversion is enabled by cantilevered beam attachments undergoing large-angle out-of-plane oscillations. By analyzing the unit cell of this locally-resonant plate, we interpret the transverse-longitudinal coupling as a nonlinear, anisotropic effective mass density. The mode conversion effect is demonstrated via direct numerical simulations. [Work supported by NSF and ARL:UT.]

10:15

2aSA8. Ultra-open acoustic metamaterial silencer. Reza Ghaffarivardavagh, Jacob Nikolajczyk (Mech. Eng., Boston Univ., 8 Saint Mary St., Photonic Bldg., Rm. 832, Boston, MA 02215, ghaffari@bu.edu), Stephan Anderson (Boston Univ. Medical Ctr., Boston, MA), and Xin Zhang (Mech. Eng., Boston Univ., Boston, MA)

Recently, with advances in acoustic metamaterial science, the possibility of sound attenuation using subwavelength structures, while maintaining permeability to air, has been demonstrated. However, the ongoing challenge addressed herein is the fact that among such air-permeable structures to date, the open areas represents only small fraction of the overall area of the material. In the presented work, in order to address this challenge, we firstly demonstrate that a transversely-placed bilayer medium with large degrees of contrast in the layers' acoustic properties exhibits an asymmetric transmission, similar to the Fano-like interference phenomenon. Next, we utilize this design methodology and propose a deep-subwavelength acoustic metasurface unit cell comprising nearly 60% open area for air passage, while serving as a high-performance selective sound silencer. Finally, the proposed unit cell performance is validated experimentally, demonstrating a reduction in the transmitted acoustic energy of up to 94%. This ultra-open metamaterial (UOM) design, leveraging a Fano-like interference, enables high-performance sound silencing in a design featuring a large degree of open area, which may find utility in applications in which highly efficient, air-permeable sound silencers are required, such as smart sound barriers, fan or engine noise reduction, among others.

10:30

2aSA9. Design, optimization, and fabrication of mechanical metamaterials for elastic wave control. Timothy F. Walsh (Computational Solid Mech. and Structural Dyn., Sandia National Labs., P.O. Box 5800, MS 0380, Albuquerque, NM 87185, tfwalsh@sandia.gov)

Harsh shock and vibration environments are commonly encountered in engineering applications involving dynamic loading. Acoustic/elastic metamaterials are showing significant potential as candidates for controlling wave propagation and isolating sensitive structural components. However, these materials have complex microstructures that must be properly designed to achieve their desired properties. In this talk we will present strategies for PDE-constrained design optimization of locally resonant elastic/acoustic metamaterials. We will present a variety of resonator geometries that can be easily optimized for wave control applications, along with fabrication details involving multi-material additive manufacturing. A variety of objective functions will be compared for their effectiveness in designing mechanical filters. Numerical examples will be presented for vibration isolation and acoustic cloaking applications. Sandia National Laboratories is a multimission laboratory managed and operated by National Technology and Engineering Solutions of Sandia, LLC, a wholly owned subsidiary of Honeywell International, Inc., for the U.S. Departmentof Energy's National Nuclear Security Administration. With main facilities in Albuquerque, NM, and Livermore, CA, Sandia has major R&D responsibilities in national security, energy and environmental technologies, and economic competitiveness.

10:45

2aSA10. Measuring anisotropy in underwater inertial metamaterials. Colby W. Cushing, Preston S. Wilson, Michael R. Haberman (Appl. Res. Labs and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, colby.cushing@utexas.edu), Chen Shen, and Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC)

The acoustic behavior of acoustic metafluids, designed and fabricated for underwater applications has been studied experimentally. Unit cells consist of elastic elements coupled via compliant layers to produce negligible shear modulus and anisotropic dynamic effective density and phase speed in orthogonal directions. Finite element simulations were used to design a unit cell that exhibits subsonic phase speeds in one direction and supersonic speeds in the orthogonal direction. Numerous samples were constructed to experimentally validate these predictions. Various combinations of materials were employed to enhance anisotropy, simplify construction, and reduce unwanted effects. Uniformly-spaced samples were tested from 0.05-10 kHz in a one-dimensional, resonator tube filled with degassed water. An electrodynamic shaker excited the system with frequency-modulated chirps. The system response was recorded using a hydrophone positioned near the top of the tube, and the resonance frequencies were used to infer the phase speeds for each mode. To extract material properties from these measurements, an effective medium model was used to represent the material-water mixture and an exact analytical dispersion relation was used to correct for elastic waveguide dispersion. Extracted material properties indicated anisotropy was achieved, and were found to be in good agreement with the as-designed properties. [Work supported by ONR.]

11:00

2aSA11. Experimental demonstration of an underwater acoustic leaky wave antenna. Michael Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 1, Austin, TX 78758, mikelee@arlut.utexas.edu), Curtis Wiederhold (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Kyle Spratt (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Christina J. Naify (none, Sacramento, CA), and Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Low-power acoustic imaging instrumentation can have a significant impact on underwater exploration and monitoring by enabling longer duration missions. Acoustic leaky wave antennas (LWAs), which consist of a dispersive analogue aperture coupled to a single electro-mechanical transducer, are a promising technological solution to address the demanding requirements of low-power underwater imaging. The dispersive nature of an LWA permits steering an acoustic beam from forward-to-backward directions by simply changing the frequency of the electro-mechanical transducer when used as either an acoustic source or receiver. While acoustic LWAs have been shown to achieve imaging resolution on par with active phased arrays, to date they have only been demonstrated for air-borne acoustic waves. This work presents the first realization of an underwater acoustic LWA using a leaky elastic waveguide consisting of a finite array of locally resonant elastic structures coupled to a single piezoelectric transducer. Finite element analysis was used as a design tool, and preliminary measurements in an underwater testing facility demonstrate forward-to-backward radiated beam scanning of a periodic, heterogeneous aluminum antenna. [Work supported by the Strategic University Research Partnership at NASA-JPL.]

11:15

2aSA12. Measurement of additively manufactured anisotropic three-dimensional underwater pentamode materials. Colby W. Cushing, Preston S. Wilson, Michael R. Haberman (Appl. Res. Labs and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, colby.cushing@utexas.edu), Xiaoshi Su, and Andrew Norris (Dept. of Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

Pentamode (PM) materials are three-dimensional (3D) elastic lattice materials that can be designed to match the acoustic impedance of water while also minimizing shear modulus of the sample over wide frequency ranges. Further, the lattice structure provides the degrees of freedom necessary to produce strongly anisotropic stiffness that is useful for transformation acoustics [Su *et al.*, *J. Acoust. Soc. Am.* **141**(6) (2017)]. In the current work, anisotropic sub-wavelength 3D metallic PM samples were fabricated using additive manufacturing for measurement of their effective material properties. The samples were distributed uniformly in a one-dimensional, water-filled resonator tube. An electrodynamic shaker excited the system and the system response was recorded using a hydrophone positioned near the top of the tube. The resulting resonance frequencies were then used to infer the phase speeds for each mode in the fluid-filled elastic waveguide. Two effective medium models were used to infer the PM-water mixture properties: (i) Wood-Mallock mixture law and (ii) a self-consistent micromechanical model that explicitly considers material anisotropy. Experimental results are compared with simulations and observations are drawn on material property extraction using the resonance tube technique when elastic anisotropy is present. [Work supported by ONR.]

Session 2aSCa

Speech Communication: Perception of Speech Directed Toward Infants and Children

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Chair's Introduction—8:00

Invited Papers

8:05

2aSCa1. ManyBabies1: Infants' preference for infant-directed speech. Melanie Soderstrom (Psych., Univ. of Manitoba, 190 Dysart Rd., Winnipeg, MB R3T 2N2, Canada, m_soderstrom@umanitoba.ca)

There is growing concern about the replicability of basic findings in psychology, including in infancy research (Frank *et al*., 2017). ManyBabies is a large-scale international collaboration to replicate basic empirical findings in infancy. Our first project is ManyBabies1, which examines infant preference for infant-directed speech (IDS). Over 70 laboratories collaborated to collect data from over 2500 infants aged 3–15 months. Stimuli consisted of speech produced in a semi-naturalistic elicitation task where fifteen different mothers who spoke North American English talked about a series of novel and familiar objects to their infant and separately to an experimenter. A set of 8 passages each in IDS and adult-directed speech (ADS) were created after a comprehensive norming process. Infants were tested in 3 primary methods: eyetracking, central fixation and headturn preference. Overall, the effect of preference for IDS was replicated, although the calculated effect was smaller than that reported by meta-analysis (Dunst *et al*., 2012, metalab.stanford.edu). Preference for IDS increased across development. The success of this project shows that infant research conducted collaboratively and at-scale can answer new questions about the replicability and generalizability of infant findings across different laboratory, methodological, and infant participant characteristics.

8:25

2aSCa2. ManyBabies1 part 2: Influences of language experience on infant-directed speech preference. Melanie Soderstrom (Psych., Univ. of ManitobaMB, 190 Dysart Rd., Winnipeg, MB R3T 2N2, Canada, m_soderstrom@umanitoba.ca) and Krista Byers-Heinlein (Psych., Concordia Univ., Montreal, QC, Canada)

ManyBabies1, our first effort at a large scale collaborative infant experimental study, provided a conceptual replication of the well-known phenomenon of infant preference for the characteristics of Infant-directed speech (IDS). One important question that has largely been unanswered by extant literature is how much the IDS preference is dependent on experience with a specific language. How do infants respond to IDS that is in a non-native variety, and how does their listening affect this preference? ManyBabies 1 used a consistent stimulus set of North-American English (NAE), which allowed us to answer this questions using two approaches. First, because participating ManyBabies 1 labs were located around the world, we were able to compare monolingual infants from a range of native-language backgrounds. We found that the preference for North American English IDS was larger for infants whose native language was NAE than for infants who had a different native language. Second, we conducted a sister project, ManyBabies 1 Bilingual, which tested infants from a variety of bilingual backgrounds. Bilinguals have similar total language experience and maturation as monolinguals, but their experience is divided across two or more languages. Planned analyses will examine monolingual-bilingual differences, and "dose-response" effects of exposure to NAE.

8:45

2aSCa3. Infant-directed speech facilitates neural encoding of speech during infants' first year of life. Marina Kalashnikova (The Basque Ctr. on Cognition, Brain and Lang., Paseo Mikeletegi 69, San Sebastian, Guipuzkoa 20009, Spain, m.kalashnikova@bcbl.eu)

Infant-directed speech (IDS) is the special speech register that parents use when addressing young infants. Compared to adult-directed speech (ADS), it is characterised by positive affect, high pitch and wide pitch range, and acoustically exaggerated speech sounds. These acoustic components of IDS have been proposed to facilitate early language acquisition by attracting infants' attention to speech, or by providing them with a type of speech input that is easier to perceive and learn, or both. Indeed, infants are more successful in a variety of behavioural language processing tasks when presented with IDS stimuli, but it is unclear whether these effects are due to the linguistic function of IDS or solely to its attention-grabbing qualities. I will present three studies that address this question by directly

measuring infants' neurophysiological responses to IDS and ADS. Our findings demonstrate that IDS elicits greater neural activation and more mature neural response patterns compared to ADS in measures of neural entrainment to continuous speech, phonemic discrimination, and semantic processing in 4-to-9-month-old infants. This evidence indicates that beyond capturing infants' attention to speech, IDShas a privileged status in facilitating neural encoding of speech, which may augment infants' early speech processing and even later language development.

9:05

2aSCa4. Effects of hearing loss and amplification device on infants' perception of infant-directed speech. Tonya Bergeson (Commun. Sci. and Disord., Butler Univ., 4600 Sunset Ave., JH246, Indianapolis, IN 46208, tbergeso@butler.edu), Yuanyuan Wang, and Derek Houston (Otolaryngol., The Ohio State Univ., Columbus, OH)

Caregivers typically speak to their infants in a speech register known as motherese or infant-directed speech (IDS; Fernald and Simon, 1984; Snow, 1977). Infants also prefer to listen to IDS over adult-directed speech (ADS) (Cooper and Aslin, 1990; Fernald, 1985; Singh *et al*.). Moreover, mothers adjust their IDS to factors such as infant age (e.g., Kitamura and Burnham, 2003; Newman and Hussain, 2006). Another factor that could potentially affect mother-infant interactions is infant hearing loss. Studies have shown that mothers adjust their speech based on infants' hearing experience (e.g., Bergeson *et al*., 2006; Kondaurova and Bergeson, 2011; Wieland *et al*., 2015). Do infants with hearing loss pay attention to IDS? Two recent studies suggest that infants who use hearing aids or cochlear implants demonstrate increased attention to IDS over both ADS and silent trials (Wang*et al*., 2017; 2018). One feature that might regulate infant attention, namely target word repetition, is correlated with vocabulary skills in later childhood (Wang *et al*., in preparation). In this talk, we will address infants' attention to speech and the role IDS serves in both cognitive-social and linguistic development for typically hearing infants and those with hearing loss.

9:25

2aSCa5. Daylong acoustic recordings of family and child speech using the HomeBank database. Mark VanDam (Speech & Hearing Sci., Washington State Univ., P.O. Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu), Paul De Palma (Comput. Sci., Gonzaga Univ., Spokane, WA), Melanie Soderstrom (Psych., Univ. of Maintoba, Winnipeg, MB, Canada), Marisa Casillas (Max Planck Inst. for PsychoLinguist, Nijmegen, The Netherlands), Alex Cristia (Laboratoire de Sci. Cognitives et Psycholinguistique, Département d'études Cognitives, PSL Res. Univ., Paris, France), Elika Bergelson (Psych. & Neurosci., Duke Univ., Durham, NC), Anne Warlaumont (Commun., Univ. of California, Los Angeles, Los Angeles, CA), Daniel Olds (Comput. Sci., Washington State Univ., Spokane, WA), and Brian MacWhinney (Psych., Carnegie Mellon Univ., Pittsburgh, PA)

HomeBank (https://homebank.talkbank.org/) is an online database of multi-hour, naturalistic audio recordings of child and family everyday experiences. Corpora in the database include (1) raw audio recordings; (2) corpus metadata including for example social details, standardized test scores, disability reports and status, family data such as number and quantity of siblings, orthographic transcriptions, output of diarization or automatic-speech recognition processing; and (3) tools for analyzing the data (https://github.com/homebankcode/) in a variety of domains. There are currently over a dozen corpora representing over 1100 multi-hour (often daylong) audio recordings. Use of the database is increasing in the scientific community, including researchers of speech, language, computer science, digital signal processing, automatic speech recognition, health sciences, and human development. The utility and extensibility of HomeBank is demonstrated in this talk with several current and ongoing projects that make critical use of the data. We discuss diarization and automatic speech processing techniques, speech and language use in pre-industrial families and groups, early perception and processing in infants, family speech and language dynamics in families of children with disorders, and database management. We are actively soliciting both users and contributors to the database.

Session 2aSCb

Speech Communication and Psychological and Physiological Acoustics: Acoustic Phonetic Properties of Speech Directed Toward Infants and Children

Mark VanDam, Cochair
Speech & Hearing Sciences, Washington State University, P.O. Box 1495, Spokane, WA 99202

Laura Dilley, Cochair
Department of Communicative Sciences, Michigan State University, East Lansing, MI 48824

Chair's Introduction—10:15

Invited Papers

10:20

2aSCb1. Understanding acoustic-phonetic environments of prelingual children with cochlear implants: Challenges, tools, and insights. Laura Dilley (Dept. of Communicative Sci., Michigan State Univ., East Lansing, MI 48824, ldilley@msu.edu)

Variability in auditory and linguistic environments experienced by prelingual children with cochlear implants (CIs) potentially helps explain variation in their language outcomes. Here, results are presented from several studies from our lab investigating early acoustic-phonetic environments of children with CIs. One collaborative study conducted over 10 years investigated individual differences in acoustic-phonetic quality of maternal speech as predictors of language outcomes in children with CIs. Results showed that properties of maternal speech recorded in the lab—as indexed in part by measures derived from vowel formants and fundamental frequency—significantly predicted infants' language outcomes and growth on multiple standardized assessments two years after cochlear implantation. Other work has examined the usefulness of a widely adopted commercial automatic speech processing technology—the Language ENvironment Analysis (LENA) system—for investigating individual differences in auditory environments. Results from our study suggested considerable variability across samples in LENA's accuracy at identifying adult speech, limiting LENA's value for investigating individual differences in acoustic-phonetic input to children with CIs. Finally, an update is provided on current work using combined signal processing and hand-coding approaches aimed at investigating variability in acoustic input to children with CIs in their home environments. [Work supported by NIH grant R01DC008581.]

10:40

2aSCb2. Using naturalistic paradigms to study how adult speakers accommodate infant listeners' unique processing demands. Elise A. Piazza, Marius Catalin Lordan, Liat Hasenfratz, Uri Hasson, and Casey Lew-Williams (Princeton Univ., 238E PNI, Princeton, NJ 08540, epiazza@princeton.edu)

Communication is inherently social and requires an efficient exchange of complex acoustic cues between individuals. What are the behavioral and neural processes that allow young listeners to understand, couple to, and learn from adult speakers in complex, everyday interactions? In one study, we recorded mothers' natural speech during play and reading and uncovered a pervasive timbre fingerprint of infant-directed speech (IDS) that generalized across 10 diverse languages. Classification of IDS and adult-directed speech (ADS) was driven by a statistical summary measure that concisely describes the vocal spectrum and could not be explained by pitch alone. In a second study, using dual-brain functional near-infrared spectroscopy (fNIRS), we measured the real-time neural dynamics of communication between infants and adults during natural interaction. We found that the infant prefrontal cortex (PFC) tracked several communicative cues, including the adult's pitch variability, with high temporal precision. Furthermore, infant-adult neural coupling was significantly greater when the members of a dyad interacted with each other than when they performed control tasks. Surprisingly, PFC activation in the infant brain slightly preceded similar activation in the adult brain, which crucially advances our understanding of children's influence over the accommodative behaviors of the caregivers around them during everyday communication.

11:00

2aSCb3. Child-directed speech in noise: Listener- and environment-related changes in speech acoustics. Nicholas A. Smith (Dept. of Speech, Lang. and Hearing Sci., Univ. of Missouri, Columbia, MO 65211, smithnich@health.missouri.edu), Christine A. Hammans, Timothy J. Vallier (Boys Town National Res. Hospital, Omaha, NE), and Bob McMurray (Dept. of Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Talkers adapt their speech in various ways according to the demands of their listeners and the communicative context. Mothers and their preschool children participated in a real-time interactive speech production/perception paradigm, in which mothers instructed their children (or an adult listener), to select the picture corresponding to the target word. The task was performed at low and high levels of background noise (56 and 76 dB SPL, delivered through headphones), to examine the effects of decreased audibility on speech production. Acoustic-phonetic analyses of child-directed speech (CDS) and adult-directed speech (ADS) productions of target words and carrier phrase (e.g., “Find pig”), revealed that mothers significantly enhanced the suprasegmental properties (i.e., pitch, intensity, and duration) of target words in CDS and at higher noise levels, but provided limited evidence for the hyperarticulation of the segmental properties of speech (i.e., formant frequencies of vowels, or voice-onset times of stop consonants). Results suggest that while some aspects of articulatory control are readily amenable to change as function of task/listener demands, others may not be. Understanding these capacities and constraints in the talking caregiver is relevant to theories of hyperarticulation in infant-directed speech.

11:20

2aSCb4. Temporal coordination of vocal turn taking between mothers and their children with hearing loss. Maria V. Kondaurova (Dept. of Psychol. & Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 46292, maria.kondaurova@louisville.edu), Nicholas A. Smith (Dept. of Speech, Lang. and Hearing Sci., Univ. of Missouri – Columbia, Columbia, MO), Qi Zheng (Dept. of Biostatistics, Univ. of Louisville, Louisville, KY), Jessa Reed (Dept. of Otolaryngology-Head and Neck Surgery, The Ohio State Univ. Medical Ctr., Columbus, OH), and Mary K. Fagan (Dept. of Commun. Sci. and Disord., Crean College of Health and Behavioral Sci., Chapman Univ., Orange, CA)

Normal-hearing (NH) infants participate in social exchanges soon after birth. What does vocal turn-taking look like in children with hearing loss after cochlear implantation? The study examined the prevalence and temporal structure of vocal turns during spontaneous interactions between mothers and their children with cochlear implants (CIs) over the first year after implantation compared to interactions between mothers and children with normal hearing. Mothers’ play with children with CIs (n = 12) were recorded at 3 (mean age 18.3 mo) and 9 (mean age 27.5 mo) months post CI. Mothers with age-matched hearing children (n = 12) were recorded at the corresponding time points. The CI group initially differed from the NH group in several ways (i.e., fewer vocal turns, more simultaneous speech, and longer between-speaker pauses) but progressed to NH levels by 9 months post CI, demonstrating the positive effects of CIs. Dyadic effects were also observed in the timing of mothers’ responses, which were related to those of their children. However, children with CIs continued to show an atypical pattern in the relative timing of between- vs within-speaker pauses across both test sessions, indicating a potentially protracted time course for the influence of CIs on dyadic interactions.

11:40

2aSCb5. Developmental cascades in reciprocal vocal signaling between infant and caregiver in typical development and autism. Gordon Ramsay, Shweta Ghai, Mitra Kumareswaran, Morgan Edwards (Dept. of Pediatrics, Emory Univ. School of Medicine, Marcus Autism Ctr., 1920 Briarcliff Rd. NE, Atlanta, GA 30329, gordon.ramsay@emory.edu), and Jhonelle Bailey (Dept. of Psych., Univ. of Miami, Coral Gables, FL)

Adults habitually adopt a special vocal register when talking to children. Current evidence suggests that the acoustic properties characterizing infant-directed speech change over development, in response to changes in infant vocal behavior, but the factors driving these changes remain largely unknown. The goal of this study is to elucidate developmental progressions in the acoustic structure of infant-directed speech over the first two years of life, and to determine the origin of these changes in infant vocal response. Samples of adult- and infant-directed speech and infant vocalizations were extracted from audio recordings of 10 typically developing infants and 10 infants later diagnosed with autism and their mothers, collected from 0 to 24 months using LENA technology. Multitaper analysis was used to determine the time-varying harmonic structure of each utterance, deriving indices summarizing differences in source and filter properties between infant- and adult-directed speech. In typical development, transitions were found towards the end of the first year of life from baby talk, emphasizing prosodic properties, towards mature child-directed register, emphasizing resonance dynamics. In autism, caregivers persisted in baby talk or shifted into adult-directed register, concurrent with disruptions of vocal contingency, which may be stimulating the transition.

Session 2aSPa

Signal Processing in Acoustics: Beamforming, Detection and Localization

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

Contributed Papers

8:00

2aSPa1. Performance weighted blending of nested array processors. Jeff Tucker (ECE, George Mason, 1511 Black Eyed Susan Ln., Vienna, VA 22182, jtucke16@gmu.edu) and Kathleen E. Wage (ECE, George Mason, Fairfax, VA)

Nested arrays consist of two uniform subarrays: a short aperture array with half-wavelength spacing and a long aperture array with greater than half-wavelength spacing. Two approaches for estimating the scanned response require computing either the product or the minimum of the subarray responses in each look direction. In multi-source environments, the multiplicative and min scanned responses may be corrupted by cross terms [Wage, *Acoustics Today* (2018)]. When the sources are uncorrelated, snapshot averaging is often used to mitigate cross term interference. The min processor's response can contain cross terms that do not decay with snapshot averaging. Alternatively, the multiplicative processor's response contains cross terms that decay, though large numbers of snapshots may be required for the cross terms to fall below the level of those in the min processor. This talk proposes a performance weighted combination of the two processors based on Buck and Singer's (IEEE, 2018) blended dominant mode rejection beamformer. The same performance weighting function can be used to combine the nested multiplicative and min processors. The resulting universal processor adjusts the weighting in each look direction as the number of snapshots increases to favor the multiplicative processor as its cross terms average out. [Work supported by ONR]

8:15

2aSPa2. Post-processing for underwater active detection and experimental results. Lin Ma, Anbang Zhao, and Chunsha Ge (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Harbin, HLJ 150001, China, malin@hrbeu.edu.cn)

In practical engineering, the target echo signals are commonly corrupted by noise and strong interferences that severely degrade the active detection performance. In the presence of noise and interferences in the detection results, the target trajectory cannot be easily recognized. A post-processing method is proposed to cancel the noise and interferences and to make the target trajectory better to be characterized. In accordance with the difference between noise and signal, a width discriminator is used to filter out some fake peaks appeared in the detection output. Considering the environmental effects, the format-based method is used to reduce the stationary interferences, for instance, reverberation due to rocks on the bottom. The post-processing method is employed and investigated in a target underwater detection experiment. The experimental results indicate that the post-processing is effective in active detection for an engineering application.

8:30

2aSPa3. Use of frequency-difference beamforming for scatterer localization. Alexander S. Douglass, Kely J. Markley, and David R. Dowling (Mech. Eng., Univ. of Michigan, 2010 AL, 1231 Beal, Ann Arbor, MI 48109, asdougl@umich.edu)

Acoustic fields interacting with discontinuities or environmental variations lead to secondary scattered fields superimposed onto incident fields. As a result, the signal measured at a remote receiving array will be modified by the scatterer's presence. Prior work has shown that the effects of strong random scattering are reduced using frequency-difference beamforming to downshift the analysis to a below-band frequency known to be less impacted by scattering. In some applications, information about a scatterer is desired, such as location, shape, or composition. However, a relatively weak scattered field from a single scatterer is difficult to detect when the incident field dominates the measurements, as both conventional and frequency-difference beamforming ambiguity surfaces primarily provide the incident field's source information. Here, a subtraction-based algorithm is implemented with frequency-difference beamforming to locate the position of a single scatterer near a source. Simulations and water tank experiments with a 110 kHz center frequency signal are considered for incident and scattered fields with a 4 cm diameter spherical scatterer, where the scattered field's received energy is roughly 2% that of the incident field. The ability to downshift the frequency demonstrates a more robust implementation of subtraction-based algorithms for locating weak scatterers without prior location knowledge. [Work supported by ONR.]

8:45

2aSPa4. Application of adaptive line spectrum enhancer in adaptive matched filter. Xiaoliang Zhang (Systems Eng. Res. Inst. of CSSC, Beijing, China), Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin Eng. University, Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn), JiangQiao Li, Huangpu Li (Systems Eng. Res. Inst. of CSSC, Beijing, China), and Xianzhong Bu (Underwater Acoust., Harbin Eng. Univ., Harbin, China)

The most classical detector of active sonar and radar 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. In addition, the TRC-IS preprocessing method works well in an actual noisy ocean environment.

9:00–9:15 Break

9:15

2aSPa5. Statistical characterization of cross terms in snapshot-averaged multiplicative processors. Vaibhav Chavali and Kathleen E. Wage (Elec. Eng., George Mason Univ., 4217 University Dr., Fairfax, VA 22030, vchavali@gmu.edu)

Multiplicative processors generate spatial spectrum estimates for sparse arrays, such as nested and coprime arrays, by multiplying the beamformed outputs of two interleaved subarrays. Nested and coprime arrays achieve significant sensor savings compared to dense Uniform Linear Arrays (ULAs) since one or both subarrays are undersampled. Chavali *et al.* show that careful design of the subarrays and proper selection of beamformer weights can guarantee power pattern performance (response to single planewave source) comparable to a conventional ULA processor [*JASA* (2018)]. The multiplicative processors' response to multiple sources contains cross terms that appear as erroneous sources in the spectral estimate. The height of cross term peaks in the spectrum depends on the subarray design, beamformer weights, and signal powers. Ksienski and Pedinoff show that averaging the multiplicative processor output over snapshots reduces the power of uncorrelated cross terms [*IEEE* (1962)], though they do not explore how much averaging is required. This talk derives the statistics of the averaged cross terms assuming complex Gaussian planewave signals and noise. The statistical model is used to predict the number of snapshots required to eliminate uncorrelated cross term peaks from the multiplicative spectra. Analytical predictions show excellent agreement with planewave simulations. [Work supported by ONR.]

9:30

2aSPa6. Nested sensor array extension factors required to match the peak sidelobe height of a uniform linear array. Hossam Elsaadawy, Kayla M. Houte, Camille LeBlanc, James M. Slezak, and Kaushallya Adhikari (Elec. Eng., Louisiana Tech Univ., 501 Don Raneau Dr. #10348, Ruston, LA 71272, kmh074@latech.edu)

Nested sensor arrays (NSAs) are a type of non-uniform linear array (NULA) that reduce the total number of sensors used for a given aperture and retain the same resolution of a uniform linear array (ULA) [Pal and Vaidyanathan, 2010]. The peak sidelobe (PSL) height fails to fall below the acceptable standard of -13 dB associated with ULAs for basic NSA configurations. The basic NSA can be extended by a factor to improve the PSL height. Coprime sensor arrays, a related NULA design, have established sensor extension factors that have been found to reduce the PSL height to this -13 dB standard [Adhikari *et al*., 2014]. For NSAs, such an extension factor has not been established. This research finds the parameters required to extend an NSA to adequately reduce the PSLs. NSAs can significantly reduce the number of sensors required to yield a satisfactory beampattern, as low as twenty-five percent of the total aperture for some cases. Considering various array apertures, certain configurations successfully use a distinctly small aperture but require a higher percentage of those to be active. To verify our results, we construct a sixty-four sensor microphone array with which we test the optimal configurations. [Work supported by Louisiana Tech University.]

9:45

2aSPa7. Multiplicity of coprime pairs for extension of coprime sensor arrays. Pablo Johnson, Daniel Sartori, Tyler Trosclair (Elec. Eng., Louisiana Tech Univ., P.O. Box 5301, Ruston, LA 71272, pkj006@latech.edu), John Willis (Elec. Eng., Louisiana Tech Univ., Saint Francisville, LA), and Kaushallya Adhikari (Elec. Eng., Louisiana Tech Univ., Ruston, LA)

This research finds the optimum period of a basic coprime sensor array (CSA) needed to reduce the PSL height to -13 dB for both minimum and product processing methods. Our design does not have the constraint that the subarray lengths be one integer apart unlike in [Adhikari *et al*., 2014]. We apply MUSIC algorithm to the optimized CSA and compare the PSD estimate obtained by Vaidyanathan and Pal's covariance matrix estimation using a single sensor pair versus our covariance matrix estimated using all available sensor pairs. We show that minimum processing requires fewer periods than product processing to reach the PSL height of -13 dB. Larger coprime pairs require fewer periods and cause an increase in variance and a decrease in the number of lags available between the two subarrays. This decrease lowers the accuracy of the covariance matrix estimation. However, using all available sensor pairs to generate the covariance matrix greatly increases the accuracy compared to using a single sensor pair. When only one signal is present, both covariance estimates produce a similar, accurate DOA, but as the number of signals increases, our covariance estimate performs better. [Work supported by Louisiana Tech University.]

10:00

2aSPa8. Improving autonomous vehicle safety—Using acoustic source localization to influence the decision making capabilities of an autonomous vehicle. Digno Iglesias, Anthony Matriss, Akin Tatoglu, and Eoin A. King (Mech. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, eoking@hartford.edu)

The manner in which autonomous vehicles (AVs) and other vehicles will coexist, and communicate with one another, is still unclear, especially during the prolonged period of mixed vehicles sharing the road. During this time, it will be important that AVs are equipped to respond to acoustic alerts in urban environments. Consider the case of an emergency vehicle in a city. In this situation, a visual perception system alone will not suffice, as it requires a direct view of the source. The goal of this ongoing project is to define a new set of audio-visual detection and localization tools to identify the location of an unexpected rapidly approaching emergency vehicle. In particular, this paper focuses on the localization of a sound source in a complex environment by combining a ray tracing approach with a direction-of-arrival algorithm. This algorithm reports a number of source directions, arising from multiple reflections in the environment. Results are combined with a three-dimensional map, acquired live from a mobile robot equipped with digital sensors including LiDAR, and a reverse ray tracing approach is used to triangulate the likely position of the source.

10:15

2aSPa9. Distributed sensing for acoustic source localization in indoor reverberant environments. Angela Bertolino, Pratik Gandhi, Arielle Joasil, Chester Obi, Kavitha Chandra, and Charles Thompson (ECE, UMASS, CACT fa203, Lowell, MA 01854, angela_bertolino@student.uml.edu)

The acoustic source localization problem relies on the estimate of the difference in time-delays of the signal received between a pair of acoustic sensors separated by a fixed distance. In the ideal case of a room with no reflections, one can identify the intersection of constant power level maps generated by multiple pairs of sensors to isolate the location of the source. The presence of reverberation however produces additional potential source locations that create uncertainty in the source location. In this research, impulse responses are simulated for a rectangular room using an image source model that incorporates frequency dependent absorption coefficients. These spatio-temporal impulse responses are applied in conjunction with room noise to simulate the signals recorded across a two-dimensional distribution of sensor pairs. The number of sensor-pairs and their spatial distribution that can optimally predict the source location in the presence of reverberant features is discussed.

Meeting of the Standards Committee Plenary Group
to be held jointly with the meetings of the

ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for:

ISO/TC 43, Acoustics,
ISO/TC 43/SC 1, Noise,
ISO/TC 43/SC 3, Underwater acoustics,
ISO/TC 108, Mechanical vibration, shock, and condition monitoring,
ISO/TC 108/SC 2, Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles, and structures,
ISO/TC 108/SC 4, Human exposure to mechanical vibration and shock,
ISO/TC 108/SC 5, Condition monitoring and diagnostics of machine systems,
and IEC/TC 29, Electroacoustics

R. D. Hellweg, Chair, P. D. Schomer, Vice Chair, U.S. Technical Advisory Group for ISO/TC 43 Acoustics and ISO/TC 43/SC 1 Noise
Hellweg Acoustics, 13 Pine Tree Road, Wellesley MA 02482
Schomer and Associates, 2117 Robert Drive, Champaign, IL 61821

R. W. Fischer, Chair, U.S. Technical Advisory Group for ISO/TC 43/SC 3 Underwater acoustics
Noise Control Engineering, Inc., 799 Middlesex Turnpike, Billerica, MA 01821

W. Madigosky, Chair of the U.S. Technical Advisory Group for ISO/TC 108 Mechanical vibration, shock, and condition monitoring
MTECH, 10754 Kinloch Road, Silver Spring, MD 20903

M. L'vov, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles, and structures
Siemens Energy, Inc., 5101 Westinghouse Blvd., Charlotte, NC 28273

D. D. Reynolds, Chair, U.S. Technical Advisory Group for ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock
3939 Briar Crest Court, Las Vegas, NV 89120

D. J. Vendittis, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machine systems
701 Northeast Harbour Terrace, Boca Raton, FL 33431

C. Walber, U.S. Technical Advisor for IEC/TC 29, Electroacoustics
diagnostics of machine systems
PCB Piezotronics, Inc., 3425 Walden Avenue, Depew, NY 14043 2495

The reports of the Chairs of these TAGs will not be presented at any other S Committee meeting

The meeting of the Standards Committee Plenary Group will follow the meeting of Accredited Standards Committee S2, which will be held on Monday, 13 May 2019, from 5:00 p.m. to 6:15 p.m.

The Standards Committee Plenary Group meeting will precede the meetings of the Accredited Standards Committees S1, S3, S3/SC 1, and S12, which are scheduled to take place in the following sequence:

Tuesday, 14 May 2019	11:00 a.m.–12:15 p.m.	ASC S1, Acoustics
Tuesday, 14 May 2019	2:00 p.m.–3:15 p.m.	ASC S3, Bioacoustics
Tuesday, 14 May 2019	3:30 p.m.–4:45 p.m.	ASC S3/SC1, Animal Bioacoustics
Tuesday, 14 May 2019	5:00 p.m.–6:15 p.m.	ASC S12, Noise

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees and U.S. TAGs.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3, and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. Parallel Committee
ISO		
R. D. Hellweg, Jr., Chair P. D. Schomer, Vice Chair	**ISO/TC 43** Acoustics	ASC S1 and ASC S3
R. D. Hellweg, Jr., Chair P. D. Schomer, Vice Chair	**ISO/TC 43/SCI** Noise	ASC S12
R. W. Fischer, Chair	**ISO/TC 43/SC 3**, Underwater acoustics	ASC S1, ASC S3/SC 1, and ASCS12
W. Madigosky, Chair	**ISO/TC 108** Mechanical vibration, shock, and condition monitoring	ASC S2
M. L'vov, Chair	**ISO/TC 108/SC2** Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles, and structures	ASC S2
D. D. Reynolds, Chair	**ISO/TC 108/SC4** Human exposure to mechanical vibration and shock vibration and shock as applied to machines, vehicles, and structures	ASC S2
D. J. Vendittis, Chair	**ISO/TC 108/SC5** Condition monitoring and diagnostics of machine systems	ASC S2
IEC		
C. Walber, U.S. Technical Advisor	**IEC/TC 29** Electroacoustics	ASC S1 and ASC S3

2a TUE. AM

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

R. J. Peppin, Chair ASC S1
5012 Macon Rd., Rockville MD 20852

A. A. Scharine, Vice Chair ASC S1
U.S. Army Research Laboratory, Human Research & Engineering Directorate
ATTN: RDRL-HRG, Building 459 Mulberry Point Rd.
Aberdeen Proving Ground MD 21005-5425

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound, etc. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 14 May 2019.

Scope of S1: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics, including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance and comfort.

Session 2aSPb

Signal Processing in Acoustics: Acoustic Detection, Localization, and Classification (Poster Session)

Kainam T. Wong, Chair
School of General Engineering, Beihang University, 37 Xueyuan Road, Haidian District, Beijing 100083, China

All posters will be on display, and all contributors will be at their posters from 10:45 a.m. to 11:45 a.m.

Contributed Papers

2aSPb1. MVDR beamforming using one bi-axial velocity-sensor. Yang Song (Nanyang Technolog. Univ., Singapore, Singapore) and Kainam T. Wong (School of General Eng., Beihang Univ., Xueyuan Rd., Beijing, China, ktwong@ieee.org)

A bivariate velocity-sensor measured two Cartesian components of the acoustic particle velocity field, which represents the spatial gradient of the acoustic pressure field. One such bi-axial unit provides azimuth-polar two-dimensional spatial directivity. This work presents the MVDR beam-pattern for such a sensing unit.

2aSPb2. The benefit of acoustic beamforming in solving the "cocktail party problem" for persons with acquired aphasia. Sarah Villard, Christine Mason, and Gerald Kidd (Speech, Lang., & Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, svillard@bu.edu)

Acoustic beamforming has been shown to improve identification of target speech in noisy listening environments for individuals with sensorineural hearing loss. This study examined whether beamforming would provide a similar benefit for individuals with aphasia (an acquired neurological language deficit). Persons with aphasia (PWA) are known to exhibit impaired language comprehension abilities; however, most work on language comprehension in PWA has been conducted in quiet settings, and little is known about the impact of competing auditory information on performance. In this study, we measured the intelligibility of target speech masked by other speech or noise for two presentation/microphone conditions: one condition, designated "KEMAR," provided natural spatial cues via KEMAR impulse responses; the second condition, designated "BEAM," enhanced the target speech level via a single-channel beamformer. In each condition, subjects heard a target sentence at 0 deg azimuth concurrent with two independent maskers—speech or speech-shaped speech-envelope-modulated noise—from $\pm$60 deg. Threshold target-to-masker ratios were measured adaptively for each subject, with individually-determined modifications in the procedures made for the subjects. Results indicated substantially lower (better) thresholds for the beamformer condition than for KEMAR, providing preliminary evidence that PWA may benefit from the use of acoustic beamforming in complex, multiple-source listening situations.

2aSPb3. Automatic environmental soundscape classification of continuous field recordings around Lake George, NY. Mallory M. Morgan (Rensselaer Polytechnic Inst., Greene Bldg, RPI, Troy, NY 12180, morgam11@rpi.edu), Vincent Moriarty (IBM, Bolton Landing, NY), and Jonas Braasch (Rensselaer Polytechnic Inst., Troy, NY)

The amount of audio data required for long-term bioacoustics monitoring is often too large to be manually sorted. Automatic environmental sound recognition techniques are therefore applied to extract relevant acoustic stimuli and classify these stimuli after a training period. In this work, a continuous 24-h, remote audiovisual recording system was developed and deployed near a tributary stream of Lake George, NY for the automatic collection of environmental and animal sounds. Besides monitoring natural environments, this system can also be used to establish an automatic protocol for collaborative business meetings. In conjunction with the efforts of the RPI/IBM Jefferson Project to create a network of sensors continuously monitoring the lake, the goal of this research is to automatically transcribe the soundscape of the lake using a network of directional microphone arrays positioned throughout the watershed. After establishing a training database of relevant acoustic stimuli, a convolutional neural network is used to classify unprocessed audio data. Pilot studies show good results when learning takes place directly from spectrograms, since the variability of non-speech acoustic events often requires more rich detail than can be provided by MFCCs or other compressed feature sets. [Work supported by NSF #1631674, CISL, and the Jefferson Project.]

2aSPb4. Bat biosonar echo analysis using spatial audio and wideband matched-filter techniques. Hyeon Lee (Mech. Eng., Virginia Tech, 100S Randolph Hall (MC0710), 460 Old Turner St., Blacksburg, VA 24061, hlee777@vt.edu), Chen Ming (Neurosci., Brown Univ., Providence, RI), Michael J. Roan (Mech. Eng., Virginia Tech, Blacksburg, VA), and James A. Simmons (Neurosci., Brown Univ., Providence, RI)

Bats and dolphins show tremendous aptitude in hunting prey in difficult conditions. These conditions include large amounts of clutter, perhaps reverberation, and interfering signals/noisy backgrounds. Bats have been extensively studied to learn more about how they detect, track, and intercept prey. One important aspect of many of these studies is to understand the acoustics around the bat's head. In this work, a new reproduction methodology using spatial audio and matched-filter techniques will be presented. A custom-built tetrahedral 1st order soundfield microphone that captures high-frequency sound up to 80 kHz from all directions was developed to measure bat echoes in B-format. The three-dimensional echoes that the bat received, now in Ambisonic B-format were further processed using a wideband matched-filter. The matched-filter produces a wideband cross-ambiguity function (WAF) of received data and the transmitted signal for every 1 deg in azimuth and elevation. The thresholded matched-filter output provides precise estimates of bearing, elevation, range, and normal velocity. This poster will present details of the hardware and software development along with experimental results that illustrate the capabilities of this new approach.

Session 2pAA

Architectural Acoustics, Musical Acoustics, and Education in Acoustics: Higher Education Schools of Music

Brian Corry, Cochair
Kirkegaard Associates, 7733 Forsyth Boulevard, Suite 1100, St. Louis, MO 63105

Kirsten Hull, Cochair
Kirkegaard Associates, 801 W Adams St., Suite #800, Chicago, IL 60601

Chair's Introduction—1:30

Contributed Paper

1:35

2pAA1. The soundscape of music education. Gary W. Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Lucky S. Tsaih (Architecture, National Taiwan Inst. of Sci. and Technol., Taipei, Taiwan), Hyun Paek, Marylin Roa, Jennifer R. Miller, Matthew Vetterick, and Keely Siebein (Siebein Assoc., Inc., Gainesville, FL)

Research on acoustical qualities of music education spaces based in soundscape theory provides a method to link the acoustical performance of a space with the instructional tasks that occur in the space with the architectural design features of the rooms. There has been limited study of the acoustical needs of teachers and students in music education spaces. Pirn (1978) showed that music practice rooms can become too loud as multiple musicians are placed in a closed and limited room volume without sufficient sound absorbing materials. Gade (1988) defined the characteristic of Support to measure cross room reflections from ceiling and wall surfaces to allow the musicians to hear each other. Tsaih (2011) found that the soundscape of music education was based on the ability of the students and teacher to be able to clearly understand spoken instruction and for students to hear each other and for the instructor to hear students playing in time, in tune and in dynamic. Case studies of music education spaces will illustrate how these concepts are integrated within the architectural design and acoustical analysis of the spaces.

Invited Papers

1:50

2pAA2. Lewis center for the arts at Princeton University. Joe Solway, Casey Eckersley, Joseph Digerness, and Raj Patel (None, 77 Water St., New York, NY 10005, joe.solway@arup.com)

The opening of Princeton's Lewis Center for the Arts in the Fall 2017 provided much needed performance and teaching spaces for the Theatre, Dance and Music programs at Princeton University, including an acoustically isolated orchestral rehearsal room, black box theater, dance theater, dance and acting studios, and music practice rooms individually hung from the roof slab above. Achieving the right balance of aural connectivity and sound isolation was an important part of the design, ensuring the Center feels vibrant and alive, while allowing teaching, rehearsal and performance to occur without disturbance. A key component to this work was the use of the Arup Soundlab, used to simulate the predicted acoustic environment within the music, drama and dance spaces, allowing the Princeton faculty and administration to listen to these spaces, and through discussion with Arup and architects from Steven Holl and BNIM, make informed decisions on the acoustic requirements for the design.

2:10

2pAA3. Glazer music performance center at Nazareth College. David Kahn (Acoust. Distinctions, 400 Main St., Ste. 600, Stamford, CT 06901, dkahn@ad-ny.com)

The new Glazer Music Performance Center at Nazareth College opened to rave reviews in Fall 2018. This project had several unique challenges including budget, schedule and a goal for simultaneous use of a warm-up room that sits within the footprint of the upper volume of the concert hall. The concert hall, while only seating 700, has a performance platform sized for a full symphony orchestra and choir. In order to acoustically accommodate these large ensembles, the room volume was set to 625 000 cubic feet. Since this would have required an unusually and impractically tall space, additional volume was captured above some of the hall's circulation areas, backstage, and above a few acoustically sensitive warm-up, teaching and practice spaces located behind the backstage area. Substantial sound isolation construction was developed to allow for simultaneous use. A second and related challenge was the project budget of $14M, translating to an average construction cost of $580/SF and included no low-cost spaces. A third challenge was an extremely aggressive project schedule requiring the facility to open in time for their 50th anniversary celebrations. All challenges were overcome thanks to an integrated design process carried out by talented and dedicated design and construction professionals.

2:30

2pAA4. DePaul University—Holtschneider performance center. Joseph W. Myers, Kirsten Hull (Kirkegaard Assoc., 801 W. Adams St. 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com), and Brian Corry (Kirkegaard Assoc., St. Louis, MO)

The DePaul School of Music has thrived for years in spite of teaching, rehearsal, and performance facilities previous facilities that were inadequate in terms of room acoustics, sound isolation and background noise. The new Holtschneider Performance Center in Chicago IL is the first and largest element in a three-phase design that utterly transforms the School of Music's facilities, finally giving the School facilities that match the quality of the faculty and students. The new building includes a 505 seat concert hall, three recital spaces, large rehearsal rooms, classrooms, numerous practice rooms and a recording suite. With such an extensive program the main challenge was to fit the full program on an urban site while respecting zoning limits to size and height. With careful organization, heavy concrete structure and thoughtful use of isolation joints, floating slabs and resilient walls and ceilings, the building provides excellent isolation despite some challenging adjacencies. The site drove an approach in which carefully controlled footprints for the major spaces are married to generous heights.

2:50

2pAA5. A multidisciplinary music and visual arts center at Earlham College. John T. Strong and Carl Giegold (Threshold Acoust., 141 W. Jackson Blvd., Ste. 2080, Chicago, IL 60604, jstrong@thresholdacoustics.com)

The Center for Visual and Performing Arts is a new construction on the campus of Earlham College, conceived as the new performance and rehearsal home for the Earlham Music department. It hosts a broad spectrum of ensemble types, including dedicated rehearsal spaces for jazz, percussion, and Javanese gamelan, and a full teaching studio and practice room suite. The flexible 260-seat Lingle Recital Hall hosts performances of small- to medium-sized ensembles; seating retracts to provide a flat floor rehearsal space for full orchestra and other large ensembles. An acoustically diffusive basket-weave wooden slat treatment around the lower perimeter of the room as well as variable acoustic systems spread throughout the hall allow a wide range of interior acoustic flexibility. The building also hosts a black box studio theater programmed by the Theater Arts department, and painting, textile, ceramics, and metalworking space for the Art department. Due to the building's compact footprint, and the high sound levels produced by the ensembles and art workspaces, extensive structural and architectural acoustic isolation measures are utilized. A centrally located recording suite with connectivity to the major performance and rehearsal spaces also functions as a teaching space for the recording arts.

3:10–3:25 Break

Contributed Paper

3:25

2pAA6. The evolution and creation of schools for music education. Hyun Paek, Gary W. Siebein, Marylin Roa, Jennifer R. Miller, and Matthew Vetterick (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, hpaek@siebeinacoustic.com)

The creation of a school of music can begin with a vision of one person or with the necessity felt by many. Case studies of two schools of music is presented which were conceived from very different approaches. Case study one shows an evolution of a complex building that started from the use of a low ceiling classroom designed for general instruction that is now a multistory facility that houses not only spaces for the entire music program at a college in Florida but also their entire fine arts program with a dormitory that encompasses the "live/learn" concept of the school. The second case study involves a first of its kind magnet school for the arts of a school district in Georgia that had no other facility to instruct talented students of music and fine arts on a vacant site. No instructors or directors of music was available for input in the programming and concept phases of the design and thus the design was left for the design team and the school district administrators to cooperate in generating the school's vision for the future.

Invited Papers

3:40

2pAA7. University of Colorado, Colorado Springs—Ent center for the arts. Matt Nichols (Jaffe Holden, 114A Washington St., Norwalk, CT 06854, mnichols@jaffeholden.com)

The UCCS Ent Center for the Arts is a transformational five-venue center; an innovative collaboration involving the university, six community arts partners and three local school districts. Jaffe Holden provided acoustic and audio/video systems design services for a wide range of spaces including a multi-use hall, recital hall, art gallery, drama theaters, art lab, recording studio, music rehearsal room, teaching studios and practice rooms. Our acoustic focus within the Shockley-Zalabak Theater was on making a successful multi-use hall to support a wide range of programming including full orchestra ensembles and small chamber groups. The acoustic response can be adjusted by a motorized system of drapes and acoustic banners concealed above a partially sound-transparent ceiling. A stage lift allows changes to be made to seating configuration and provides a stage extension. An acoustical shell can be configured to accommodate both large and small group performance. The Chapman Foundation Recital Hall's acoustics can also be tailored to suit various musical ensembles, thanks to an adjustable acoustic curtain system hidden above the partially sound-transparent ceiling and slatted wood wall at the rear of the stage. One of the biggest challenges was to successfully isolate a dance studio directly above a recording studio control room.

4:00

2pAA8. Armerding center for music and the arts at Wheaton College. Gregory A. Miller, Marcus R. Mayell, and Dawn Schuette (Threshold Acoust., LLC, 141 W. Jackson Blvd., Ste. 2080, Chicago, IL 60604, gmiller@thresholdacoustics.com)

Wheaton College has long cultivated young musicians in a tight-knit community outside of Chicago. As the program has grown in recent decades, however, the community has burst at the seams in need of expanded space for faculty and students alike. The construction of a new science building on campus led to the availability of Armerding Hall to be redeveloped into a new home for the Conservatory on the central campus quad. Shallow floor-to-floor heights and limited structural capacities led to acoustic isolation strategies that provide "just enough" separation between faculty studios and relied on careful detailing for success. A steeply-raked Lecture Hall was reimagined to become a 100-seat recital hall with volume carved into adjacent spaces. Other spaces within the building include ensemble rehearsal rooms, a recording suite, and a room for multi-media composition. A second phase of the project will include new construction of a 650-seat concert hall and the college's first dedicated rehearsal hall for their formidable choral program.

4:20

2pAA9. Missouri State University—Ellis Hall. Brian Corry (Kirkegaard Assoc., 7733 Forsyth Blvd., Ste. 1100, St. Louis, Missouri 63105, bcorry@kirkegaard.com), Kirsten Hull, and Joseph W. Myers (Kirkegaard Assoc., Chicago, IL)

Missouri State University Ellis Hall is a 1957 modernist building originally designed for the music department. The building was mostly untouched over the years, despite poor sound isolation and unremarkable, undersized performance spaces. Kirkegaard worked closely with Patterhn Ives on a comprehensive renovation that included replacing the building mechanical systems and much of its exterior curtain wall. The interior is a virtual gut/rehabilitation, but wherever possible existing walls were incorporated into new sound-isolating assemblies to avoid unnecessary expense. The substantial scale of the project compared to the very limited budget required the design team to find carefully calibrated solutions for what to reuse and what to replace. One of the greatest challenges was to provide an acoustic superior recital hall, a goal that the School of Music emphasized as critical to the future of their program, from a renovation of the low-ceilinged existing recital hall. The solution was to demolish the hall's floor slab to capture the room volume from an under-used rehearsal room below, while preserving the upstage pipe organ and a handsome wood side wall. The result is a very successful new 240 seat recital hall in the shell of the old hall.

4:40

2pAA10. South Dakota State University. David Kahn (Acoust. Distinctions, 400 Main St., Ste. 600, Stamford, CT 06901, dkahn@ad-ny.com)

Shortly after South Dakota set out to design their new performing arts center to support their music and theatre programs in 1997, they discovered the cost to construct these facilities was roughly double their budget. After 6 years of planning and design, the phase 1 performing arts center opened. Then, after an additional 5 years, the University was able to raise enough money to construct the second phase of their performing arts center which included a state—of-the-art large proscenium theatre, rehearsal rooms for choir and large instrumental ensembles, a jewel box recital hall with a pipe organ, music teaching studios and practice rooms. The accommodation of the pipe organ into the design of the recital hall was especially challenging because the pipe organ was originally designed and installed in a much larger and different-shaped church. Pipe organ music was not a high priority for the use of the new recital hall, and therefore some special accommodations were made to improve the sound of the pipe organ without compromising the acoustics for the room's primary use as a recital hall.

Session 2pAB

Animal Bioacoustics and Signal Processing in Acoustics: Bioinspiration and Biomimetics in Acoustics II

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

Chair's Introduction—1:00

Invited Paper

1:05

2pAB1. The evolution of bat robots. Rolf Müller (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Blacksburg, VA 24061, rolf.mueller@vt.edu), Roman B. Kuc (Elec. Eng., Yale, New Haven, CT), and Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Tech, Roanoke, VA)

The extraordinary skills of bats in supporting dexterous mobility in complex environments based on just two pulsed trains of one-dimensional biosonar echoes has attracted attention from engineers for several decades already. To explore whether it is possible to reproduce at least certain of these capabilities, a diverse set of "bat robot" prototypes have been built. The earliest, most basic of these systems were limited to estimating the distance of sonar targets based on the acoustic time-of-flight. From there, systems improved to take advantage of more echo waveform features, e.g., for target recognition. Two (or more) receivers were introduced to exploit binaural differences, e.g., for target tracking. Rigid ear rotations served to enhance the signal-to-noise ratio be focusing on a target or to determine target direction from the echo amplitudes received across a scan. Biomimetic emission and reception baffle shapes, i.e., "noseleaves" and "pinna," were added to narrow the sonar beams and create direction-depended spectral signatures. Deformations of flexible baffle structures that mimic the muscular actuation of the noseleaf and pinna shapes seen in bats have been added to these systems. Mobility of the entire systems has been provided by mounting them on pan-tilt units, robot arms, mobile robots, and drones.

Contributed Papers

1:25

2pAB2. A soft robotic actuation system for a bat robot. Emmet Eckman, Matthew Conk, Sierra Lunsford, Charlie Martin, Agoshpreet Singh, Kent Sullivan, Yanhao Wang (Mech. Eng., Virginia Polytechnic Inst. and State Univ., 1075 Life Sci. Circle, Blacksburg, VA 24060, eckman19@vt.edu), Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Polytechnic Inst. and State Univ., Roanoke, VA), and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

A crucial development in the field of biomimicry is accurately recreating the kinematics of various organisms. In the example of a bat robot in development inspired by the greater horseshoe bat, *Rhinolophus ferrumequinum*, this takes the form of accurately recreating fast motions of the pinnae and noseleaf observed during echolocation. In order to accurately recreate these baffle shapes it is important to choose a material that is malleable but will retain its originally molded state. An actuation system was developed to replicate the ear movement seen in the bat when using biosonar. Several actuation designs were tested, including a physical motor connection, shape memory alloys, and soft-robotic pneumatics. In addition to the actuation system, a feedback system was developed in order to accurately control the robot and provide information which could be used to determine the effectiveness of the robot. Additionally, the final challenge to assemble all the components, i.e., the baffle shapes, actuator mechanism, and feedback control mechanism, in a way that recreates the baffle motions in an effective and accurate manner.

1:40

2pAB3. Design and manufacturing of a miniature pneumatic actuator for a bat robot. Kent Sullivan, Matthew Conk, Emmet Eckman, Sierra Lunsford, Charlie Martin, Agoshpreet Singh, Yanhao Wang (Mech. Eng., Virginia Polytechnic Inst. and State Univ., 1075 Life Sci. Circle, Blacksburg, VA 24060, kntsully@vt.edu), Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Polytechnic Inst. and State Univ., Roanoke, VA), and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

There have been many pneumatic actuator designs created over the past several years. One such design consists of multiple, completely separated air cells that when pressurized, expand and push off one another causing the actuator to bend. With this chamber design, the actuator requires a small change in volume to deform, effectively leading to improved reliability and increased actuation speed. Other design characteristics such as cross section geometry, inner/outer wall thickness ratio, distance between cells, cell height, and material selection can be modified to optimize the actuator's performance. Incorporating this design into the bat robot allowed for greater soft robotic ear deformation, however it is too large for this application. To see if it was possible to reduce the size of the actuator while maintaining its functionality, several adjustments were iteratively made to previously described mold designs. Manufacturing remained a two-step process, but removable side walls and a filter-ventilation system were added to allow for the silicone, used to cast the actuator, to fully cure. Ultimately, the actuator produced maintained its expected functionality and was produced at a smaller scale than originally thought possible.

1:55

2pAB4. Acoustic emission system for a bat robot. Nathan Cox (Aerosp. and Ocean Eng., Virginia Polytechnic Inst. and State Univ., 1075 Life Sci. Circle, Blacksburg, VA 24060, nathc17@vt.edu), Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Polytechnic Inst. and State Univ., Roanoke, VA), and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

One of the more difficult challenges in producing a biomimetic bat robot is replicating the bats' biosonar emission capabilities. For mimicking horseshoe bats (Rhinolophidae), this means creating a high-power acoustic source that can also properly illuminate the noseleaf structure over a bandwidth of approximately 20 kHz. Computer tomography (CT) scans were used to inform an accurate noseleaf model which was then abstracted from life using three-dimensional mesh software. A waveguide was designed to direct sound from two electrostatic transducers into a small nozzle, creating a point source. Numerical acoustic simulations were used to optimize these waveguide shapes to obtain a good compromise between point source behavior, high output amplitudes, and minimal internal reverberation. Physical geometries were procedurally generated to create a three-dimensional design space with parameters eccentricity, length and convexity. Data from each waveguide were analyzed for volume, multidirectionality, and echoing. A second round of simulations was executed to determine the effect of nozzle width on the output. Findings indicated that waveguide convexity and eccentricity played primarily into echoing and increased length reduced output volume. As nozzle diameter increased, the output volume increased, but sound became unidirectional. Future research may include development of alternative emission systems without a need for waveguides.

2:10

2pAB5. Bat-board: An integrated electronic system for a robotic bat. Lucas Mun (Elec. and Comput. Eng., Virginia Polytechnic Inst. and State Univ., 1075 Life Sci. Cir, Blacksburg, VA 24061, cmun09@vt.edu), Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Polytechnic Inst. and State Univ., Roanoke, VA), and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

The biosonar of horseshoe bats combines the standard elements of any sonar system, i.e., pulse generation and echo processing, with a peripheral dynamics provided by deforming baffle shapes for ultrasonic emission ("noseleaves") and reception (ears). Realizing a "bat robot" to reproduce this biological system requires the implementation of three main electronic functionalities: power distribution, analog signal conditioning, and robot control. To meet the size constraints of the bat robot, the work presented here has designed/implemented a "Bat Board" that integrates all three of these functionalities within a single circuit board. The power distribution component of the Bat Board supplies the high voltage amplifier, the air compressors/valves for the pneumatic actuators in the noseleaf/the ears of the sonar head, and the data acquisition systems. The high voltage signal amplification component was designed to power the electrostatic transducers that were selected for their broad-band output. To combat interferences originating from the power distribution and the digital control signals, different PCB layout paradigms were implemented in the design process such as separating ground planes and the extensive use of bypass capacitors. The result of this efforts has been a single PCB implementation with a very compact footprint and the potential for further miniaturization.

2:25

2pAB6. Data acquisition and controls for a bat robot. Brandon Walker (Elec. and Comput. Eng., Virginia Polytechnic Inst. and State Univ., 1075 Life Sci. Cir, Virginia Tech, Blacksburg, VA 24061, branw@vt.edu), Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Polytechnic Inst. and State Univ., Roanoke, VA), and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

Robotic reproductions of the dynamic and adaptive nature of the biosonar systems found in many bat species require a capable back-end that is capable of integrating mechanical, acoustical, electrical, and computational functions in an efficient manner. This is particularly true when mimicking bat species that change the shapes of their noseleaves and pinna as part of their biosonar behaviors. To address the challenge of replicating these highly integrated functions, a platform consisting of commercial, off-the-shelf components, centered around a microcontroller (Arduino Due) and a single-board computer (Raspberry Pi 3) has been designed. The real-time microcontroller has been delegated the signal generation for the ultrasonic pulses and data acquisition for the echoes, as well as all control operations for the mechanical periphery of the robotic bat head. This includes synthesizing output waveforms, conditioning measured data, and maintaining the state of the pneumatic actuation systems. Commands are issued directly by the computer, which is responsible for orchestrating overall "behaviors" and managing the relevant data. In addition, the control system acquires and stores meta data for the echoes such as geospatial location and acquisition time. Future improvements to this system will seek to establish closed loop control liking echo analysis to peripheral dynamics.

2:40

2pAB7. Integrating a brain into a bat robot. Mohammad Omar Khyam (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Blacksburg, VA 24061, mok@vt.edu), David Alexandre (Aerosp. and Ocean Eng., Virginia Tech, Blacksburg, VA), Ananya Bhardwaj, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

The sophisticated biosonar systems of horseshoe bats have enabled these animals to navigate and pursue prey in complex environments. A conspicuous peripheral dynamics in which the animals' noseleaves and pinnae change during biosonar behaviors could play an important role in enabling these capabilities. It may be hypothesized that for the integration between peripheral dynamics and neural signal processing/estimation to be maximally effective, the periphery should be controlled by feedback from the output of the subsequent neural echo processing. In the way, the specifics of sensory information encoding in the periphery could be controlled based on the needs of the neural signal processing. As a first step towards such an integration in a biomimetic sonar head, a computational model for the inner ear and the auditory nerve's spike code has been integrated with a dynamic periphery that—like the computational models—mimics horseshoe bats. For each model stage, alternative versions with different levels of complexity have been implemented to test how module complexity and the values of the associated parameters affect the capacity of the echo representation to encode sensory information. These effects have been tested based on a large dataset of 220 000 echoes collected in natural forest environments.

2:55

2pAB8. Sonar signal representation mimicking the inner ear of horseshoe bats. Mohammad Omar Khyam, Ananya Bhardwaj (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Blacksburg, VA 24061, mok@vt.edu), David Alexandre (Aerosp. and Ocean Eng., Virginia Tech, Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Current models of the initial stages of auditory processing in mammals usually agree that the input signals are split into a bank of bandpass filters. However, the available models differ substantially in their level of complexity and the number of parameters needed. In the current work, the varying levels of complexity in filterbank models of the basilar membrane have been evaluated in the context of modeling the cochlear processing of natural biosonar echoes in horseshoe bats. To this end, three different types of filterbank models have been implemented to represent the range of complexity spanned by models that are in use for the human inner ear: The simplest model, a gammatone filterbank, is a linear model with symmetric filter transfer functions. The gammachirp filterbank is also linear, but mimics the asymmetric transfer functions of the basilar membrane. Finally, the dual resonance nonlinear (DRNL) model adds a level-dependent behavior. Here, all three models have been adapted to the specifics of the basilar membrane of horseshoe bats which is characterized by an "auditory fovea" with exceptionally high filter qualities. The outputs of the different models have been encoded into a sequence of neural spike times before being evaluated with various information-theoretic methods.

3:10

2pAB9. Biomimetic spike representation for the encoding of sonar signals. David Alexandre (Aerosp. Eng., Virginia Tech, ICTAS II, 1075 Life Sci. Circle, Blacksburg, VA 24060, davida49@vt.edu), Mohammad Omar Khyam, Ananya Bhardwaj, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Biosonar echoes received by bats in their natural habitats are short, highly time-variant acoustic waveforms. Because of these signals properties, the way in which the bats' auditory nerve represents the echoes could be a model for how sparse neural time-code can capture salient signal features. Here, the spike-encoding of natural echoes has been studied based on a large data set containing about 220 00 echoes that were collected by hand-carrying a biomimetic robotic sonar head through forest environments. The sonar head was equipped with flexible noseleaf and pinna shapes that could deform during pulse emission/echo reception in a similar fashion to what horseshoe bats do. This peripheral dynamics was turned on for half of the recoded echoes and turned off for the other half. The echo waveforms were transformed into spike trains using two spike-generation models, each with different levels of complexity: The simplest version was a "leaky integrate-and-fire model; in the more complex version, a response kernels was added to model the refractory behavior of the neurons. The input to the spike generation models were the outputs of three different basiliar membrane models of varying complexity levels. The coding capacity of the spike trains has been evaluated using information-theoretic methods.

3:25

2pAB10. Information-theoretic evaluation of brain-inspired sonar signal representation. Ananya Bhardwaj, Mohammad Omar Khyam (Mech. Eng., Virginia Tech, ICTAS II, 1075 Life Sci. Circle, Mail Code 0917, Blacksburg, VA 24061, ananya22@vt.edu), David Alexandre (Aerosp. and Ocean Eng., Virginia Tech, Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Many bat species, such as horseshoe bats, move their external auditory periphery (noseleaf and pinnae) during emission/reception of their biosonar signals. This peripheral dynamics has already been shown to enhance sensory information encoding at the level of the analog echo waveforms. However, the bats' brains do not process any analog waveforms directly but rely on spike codes generated in the auditory nerve instead. Hence, it is desirable to evaluate the effect of the peripheral dynamics at the level of these spike codes. As input for this analysis, natural foliage echoes akin to the forest environments experienced by bats were recorded using a biomimetic robot with a dynamic periphery similar to that of horseshoe bats. The echoes were converted into neural spike trains through a signal-processing model of the bats' cochlea/auditory nerve. The effect of the dynamic periphery was then investigated using information-theoretic techniques. As a first step, entropy was estimated to quantify coding capacity. The results showed an increased entropy within signals corresponding to a dynamic periphery when compared to a static periphery. For the next steps, the variability in characteristic spike train features such as spike timing, spike rates, and spike intervals will be characterized with additional information-theoretic methods.

3:40–3:55 Break

3:55

2pAB11. Bat biosonar study using spatial audio techniques. Hyeon Lee (Mech. Eng., Virginia Tech, 100S Randolph Hall (MC0710), 460 Old Turner St., Blacksburg, VA 24061, hlee777@vt.edu), Chen Ming (Neurosci., Brown Univ., Providence, RI), Michael J. Roan (Mech. Eng., Virginia Tech, Blacksburg, VA), and James A. Simmons (Neurosci., Brown Univ., Providence, RI)

Bats have highly specialized hearing adaptations that give them target detection, localization, and interception abilities that far surpass any man-made SONAR. There is a large body of research that has studied and attempted to mimic this remarkable natural sonar system using various methodologies. In many studies, multiple microphones and time difference of arrival were used for bat ultrasonic wave measurements. In this work, a new method using spatial audio techniques is introduced for bat biosonar measurements and analyses. A custom-built tetrahedral soundfield microphone that captures high-frequency sound up to 80 kHz from all directions was developed to acquire bat ultrasonic echoes in Ambisonic B-format. The recorded sound was processed using the HARPEx (High Angular Resolution Planewave Expansion) technique that decodes the 1st order spherical harmonics of the soundfield at the microphone into two plane waves in each time-frequency bin. The use of the soundfield microphone and associated processing provides new tools for analysis of bat echoes from the perspective of the bat. This talk will present details of the hardware and software development along with experimental validation results.

Invited Paper

4:10

2pAB12. Bio-inspired generalizable-focusing wideband FM sonar. James A. Simmons and Chen Ming (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu)

The bat's inner ear and auditory brainstem register successive frequencies in FM sweeps of broadcasts and echoes with single spikes. Using the neural spectrogram of each broadcast as reference, spike latencies trace the FM sweeps for imaging echo delay by spectrogram correlation. If mutually-interfering glint reflections are present, the echo's neural spectrogram has regularly-spaced ripples caused by amplitude-latency trading. Glint-delay estimates are generated by transforming the ripple pattern from frequency to time in real time using feedforward inhibition. Several glint delay estimates can coexist in focused target images before accumulation of nulls generates too many glint estimates and blurs the image. Echo lowpass filtering, characteristic of clutter, affects a wide swath of frequencies and changes the slope of the neural spectrogram, which evokes multiple nulls and causes image blurring that suppresses the clutter. Echo Doppler shifts also change the slope of the FM sweep, and the same mechanism is capable of blurring Doppler-shifted images so that troublesome ambiguity is suppressed. By internally adjusting the neural FM sweep of the broadcast reference, the images of echoes can be refocused onto any desired location not only in range, azimuth, and elevation, but also on the range-Doppler plane. [Work supported by ONR.]

4:30

2pAB13. Accommodating constraints for modeling of biosonar processing by big brown bats. Chen Ming, James A. Simmons (Dept. of Neurosci., Brown Univ., 185 Meeting St., Providence, RI 02912, chen_ming@brown.edu), Stephanie Haro (Program in Speech and Hearing Sci. and Technol., Harvard Med. School, Boston, MA), and Jason E. Gaudette (Sensors and Sonar Systems Dept., Naval Undersea Warfare Ctr., Newport, RI)

Constraints on modeling of wideband FM biosonar are acoustic (small targets return discrete replicas of the incident sound from individual component parts), auditory (inner-ear transduction segments broadcasts and echoes into numerous, parallel bandpass channels with integration-times of ~350 μs), neural (processing retains parallel frequency bands throughout the auditory pathway that use single spikes to mark frequencies), and perceptual (bats nevertheless perceive true delays of individual glint reflections, not just spectral coloration). The bat's target model is geometric, registering the glints binaurally along the range axis. However, because closely-spaced glints are represented by their mutual interference, not by separate acoustic reflections, the ripples have to be transposed from an interference pattern back into glint delay estimates. Auditory cortical neurons selective for specific patterns of ripple come to register instead the underlying time separation of the glint reflections. The developing SCAT model of wideband FM biosonar incorporates a network for glint delay transformation that also offers a route for clutter suppression by contrasting dispersed, unfocused clutter images with tightly focused target images in range and azimuth space. The feedforward inhibition of this network preserves real-time operation for target reconstruction as well as for guidance in clutter. [Work supported by ONR.]

4:45

2pAB14. Biomimetic foliage echo simulation. Michael Goldsworthy (Comput. Sci., Virginia Tech, 155 Otey St., Rm. 323, Blacksburg, VA 24061, michaeljg@vt.edu) and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Many echolocating bat species are capable of navigating through highly cluttered environments, such as dense foliage—apparently without difficulty. Since interpreting foliage echoes must hence be key to the animals' navigation capabilities, simulating such echoes would be a major step towards understanding bat navigation. This is a difficult task, as vegetation echoes are highly complex and stochastic signals. Prior work on modeling foliage echoes frequently relied on physical approximations, since full physics models are computationally infeasible. For simulating the echo from a leaf, a round disk approximation has been used, since it's echo has a known physical solution. These disks have been distributed according to the locations of leaves in a tree using a Lindenmayer System (L-System), thus creating a full tree echo model. Successful prior work in using a machine learning approach for echo based tree classification, shows that a data driven approach may also be useful in modeling echoes. Hence, future research will be using machine learning methods, such as auto-encoders and generative adversarial networks, to simulate the echoes of leaves and trees. The echo data to be fed into these methods will be of leaves recorded in an anechoic chamber using a biomimetic sonar head.

5:00

2pAB15. Biomimetic solutions to finding passageways in foliage. Ruihao Wang and Rolf Müller (Mech. Eng., Virginia Tech, 112 Hearthstone Dr., Apt. 210, Blacksburg, VA 24060, ruihaow1@vt.edu)

Many bats species live in densely vegetated habitats. Hence, they must have evolved the ability to detect passageways through the foliage. To identify echo features that bats have at their disposal to accomplish this sensing task, a biomimetic sonar head was used to ensonify artificial foliages in the laboratory. These foliages consisted of plastic vines that were arranged to create gaps of defined width and height. The conventional sonar approach to detecting gaps between scatterers is based on the drop in energy that occurs when the sonar beam is aimed at a gap. However, the performance of such an energy-detector decrease as the sonar-beam width increases relative to the angle subtended by the gap edges. Hence, reliable detection of a narrow gap with a wide beam and/or from a distance is not feasible with this approach. Here, a machine-learning approach based on convolutional neural networks was employed to identify features in energy-normalized spectrogram that could support passageway-detection independently of echo energy. The results indicate that features other than echo energy exist in the spectrogram that by themselves support much better passageway detection performance than the energy-based reference. Work to understand the features learned by the convolutional neural network is currently underway.

5:15

2pAB16. Biomimetic sonar and the problem of finding deterministic targets in foliage. Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Polytechnic Inst. and State Univ., 3719 Parliament Rd., Apt. 22, Roanoke, VA 24014, josephs7@vt.edu) and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

The ability of (bio)sonars to find targets-of-interest is often hampered by a cluttered environment. For example, naval sonars encounter difficulties finding mines partially or fully buried within sand. Such situations pose target identification challenges are much harder than target detection and resolution problems. There are many bat species which navigate and hunt in dense vegetation and thus must be able to identify targets-of-interest within clutter. Evolutionary adaptation of the bat biosonar system is likely to have resulted in the "discovery" of features that support making distinctions between clutter and echoes of interest. The most well-established case is given by cf-fm bats that use Doppler shifts caused by the wingbeat of a flying insect prey to identify the prey in foliage. Other bat species have been shown to use a passive sonar approach that is based on unique prey-generated acoustic signals. Some of the most interesting cases can be found in bat species that are successful in finding preys that apparently does not emit any distinguishing sounds themselves and would hence limit the bats to an active-sonar approach. Such bat species could provide model species for new ways in which the target-identification problem in clutter can be solved with active sonar.

Session 2pBA

Biomedical Acoustics and Signal Processing in Acoustics: Cardiovascular Ultrasound: Imaging and Therapy II

Kevin J. Haworth, Cochair
University of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209

Jonathan A. Kopechek, Cochair
University of Louisville, 2301 S Third St, Lutz Hall, Room 400, Louisville, KY 40292

Invited Papers

1:00

2pBA1. Ultrasound-targeted microbubble destruction (UTMD)—Novel applications for microRNA delivery. Howard Leong-Poi (Cardiology/Medicine, St. Michael's Hospital, 6-044 Queen Wing, 30 Bond St., Toronto, ON M5B1W8, Canada, leong-poih@smh.ca)

Ultrasound-targeted microbubble destruction (UTMD) is a non-invasive technique for gene delivery, utilizing high power ultrasound and nucleic acid-bearing microbubbles. UTMD has been used in a variety of *in vivo* applications, including cardiac and skeletal muscle, kidney, liver, cerebral and even lung, and have been studied using many gene vectors, including plasmid, viral and small interfering RNA. The focus of gene therapy has now shifted towards small non-coding RNAs, including microRNAs (miRNA). These non-coding RNAs are important transcriptional and post-transcriptional inhibitors of gene expression that regulate the translational output of target messenger RNAs. MicroRNAs have been shown to participate in a multitude of cellular process, and their dysregulation may play an important pathophysiologic role in many different human pathologies. Coupled with their ability to specifically target particular cellular pathways, makes the possibility of exploiting miRNAs to develop therapeutic strategies extremely attractive. This presentation will focus specifically on several miRNAs with UTMD applications in (1) chronic ischemic peripheral arterial disease, (2) ischemia-reperfusion injury, and (3) abdominal aortic aneurysms (AAA). For each application, we will discuss selection of miRNA, aspects of tissue targeting *in vivo*, enhancement of therapeutic effect and potential for clinical translation.

1:20

2pBA2. Lipid-shelled microbubbles for ultrasound-triggered release of bioactive gases to treat stroke and cardiovascular disease. Christy K. Holland, Himanshu Shekhar, and Maxime Lafond (Internal Medicine, Div. of Cardiovascular Diseases and Biomedical Eng., Univ. of Cincinnati, Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

Ischemia-reperfusion-induced neurological injury is a primary cause of stroke disability. Xenon (Xe), a bioactive gas, has potential as an effective and nontoxic neuroprotectant for the treatment of ischemic stroke. Nitric oxide (NO) is a potent bioactive gas capable of inducing vasodilatory, anti-inflammatory, neuroprotectant and bactericidal effects. The goal of this work was to develop lipid-shelled microbubbles for site-specific release of Xe or NO upon pulsed ultrasound exposure. Gas-loaded microbubbles were synthesized by high-shear mixing of a lipid dispersion in a vial that contained, Xe or NO, and octafluoropropane (OFP) in combination. Attenuation spectroscopy measurements demonstrate the feasibility of 6-MHz pulsed Doppler ultrasound-triggered release of Xe or NO from microbubbles. The addition of OFP in the lipid-shelled microbubbles increased the number density, size, and stability of the microbubbles, particularly in undersaturated saline. Gas chromatography mass spectrometry was employed to measure Xe dose (127 ± 29 μl Xe/mg lipid). The payload of NO in the microbubbles (97 ± 12 μl NO/mg lipid) was assessed using an amperometric sensor. Intravenous administration of microbubbles carrying a neuroprotective or a vasodilatory gas in combination with ultrasound exposure has potential as a novel noninvasive strategy for local therapeutic delivery to modulate the effects or duration of cerebral ischemia.

Contributed Papers

1:40

2pBA3. Remote implantation of ultrasound responsive multi-cavity microparticles for early atherosclerosis. Xiaoqian Su, Reju G. Thomas, Lakshmi Deepika Bharatula, and James J. Kwan (School of Chemical and Biomedical Eng., Nanyang Technol. Univ., 62 Nanyang Dr., Block N1.2, 01-06, Singapore 637459, Singapore, jameskwan@ntu.edu.sg)

Atherosclerosis is an inflammatory disease of arteries, and results in stroke or heart attacks—the leading causes of death and disabilities in the developed world. Yet drug treatments for this chronic inflammatory disease remains to be addressed. Here, we developed a multi-cavity poly-co-lactic-co-glycolic microparticles (mc-PLGA MPs) capable of being remotely implanted at the site of arterial injury with focused ultrasound. The obtained mc-PLGA MPs present with two to five submicron surface cavities with rough inner surfaces. After exposure to high intensity focused ultrasound (HIFU) in an agarose tissue phantom, mc-PLGA MPs extravasated beyond the lumen of the channel at an average distance of 4.29 ± 1.19 mm, and sustained release of rhodamine-B for up to 15 days. Similarly, mc-PLGA MPs were able to be implanted into the sub-endothelial space of an *ex vivo* porcine artery model without observable damage to the artery. To further validate the drug release, we exposed a foam cell spheroids model to mcPLGA MPs and HIFU, and show that particles were dispersed throughout the spheroids and sustained delivery throughout. The results here highlight the potential for HIFU-guided implantation of mcPLGA MPs to improve local and sustained treatment of inflamed arterial tissue.

1:55

2pBA4. Detection of nucleic acid-loaded microbubbles in mouse hearts during ultrasound-mediated delivery. Meghan R. Campbell, Mariah C. Priddy, and Jonathan A. Kopechek (BioEng., Univ. of Louisville, 2301 S Third St., Lutz Rm. 419, Louisville, KY 40292, meghan.campbell@louisville.edu)

Heart disease is a leading cause of death worldwide. Recently, there has been a growing interest in nucleic acid-based therapeutics, such as microRNAs (miRs) or microRNA inhibitors (antimiRs), for cardiac repair. Previous studies have shown that regulating microRNA levels in the heart can promote proliferation of cardiac cells, decrease fibrosis, and improve cardiac function. Ultrasound targeted microbubble destruction (UTMD) is in development to focus the delivery of therapeutic nucleic acids to the heart and reduce adverse systemic effects. Microbubbles can be loaded with nucleic acids and exposed to ultrasound in order to induce cavitation and enhance delivery at the target site. In this study phospholipid-coated microbubbles were loaded with therapeutic miR mimics or antimiRs and injected intravenously in mice. Ultrasound pulses (2.5 MHz, 0.9 MPa peak negative pressure) were applied to the heart using a P4-1 array on a Verasonics Vantage ultrasound system. The ultrasound images were analyzed to detect microbubbles in the heart during treatment. Increased mean intensity during infusion was associated with increased delivery of nucleic acids to the heart as assessed with qPCR. The results suggest that quantitative analysis of ultrasound images to detect microbubbles *in vivo* may aid in monitoring UTMD treatment for improved cardiac health.

2:10

2pBA5. Pentagalloyl glucose effects on murine abdominal aortic aneurysms. Jennifer L. Anderson, Alycia G. Berman, Elizabeth E. Niedert (Biomedical Eng., Purdue Univ., 206 S Martin Jischke Dr., Rm. 3083, West Lafayette, IN 47907, ander934@purdue.edu), Sourav Patnaik, Ender A. Finol (Mech. Eng., The Univ. of Texas at San Antonio, San Antonio, TX), and Craig J. Goergen (Biomedical Eng., Purdue Univ., West Lafayette, IN)

An abdominal aortic aneurysm (AAA) is a dilation of the abdominal segment of the largest artery in the body and can be accompanied by significant risk of rupture and mortality. The current treatment, surgical repair, carries risks and complications. As such, there is need for less invasive therapies capable of curbing aneurysm growth. Here we use an AAA mouse model to evaluate the potential of pentagalloyl glucose (PGG) to suppress aneurysm growth. To induce aneurysms, 5.0 mg/mL pancreatic porcine elastase (PPE) was topically applied to the infrarenal aorta and 0.2% beta aminoproprionitrile was continuously administered via drinking water. Four of the eight animals also received topical treatment of 0.06% PGG via gauze before PPE treatment. High frequency ultrasound imaging (Vevo2100 system, VisualSonics) with a MS550D transducer (40 MHz center frequency) was performed prior to surgery, and every week thereafter for 4 weeks. PGG-treated animals had a small but nonsignificant decrease (p = 0.09) in effective maximum aortic diameter on days 7 and 14 that was not present on day 28. Current work is being performed to quantify PGG binding via histological characterization. Ultimately, we aim to investigate the parameters required for PGG to be an effective treatment for mechanically-stabilizing small AAAs.

2:25–2:40 Break

Invited Papers

2:40

2pBA6. Measuring thrombolysis in a static model to assess ultrasound protocol efficacy. Curtis Genstler and Misty L. Noble-Vranish (BTG plc / EKOS Corp., 11911 N. Creek Pkwy S, Bothell, WA 98011, curtis.genstler@btgplc.com)

Over the years, EKOS has developed an efficacy measure called Lysis Enhancement Factor (LEF) to evaluate the performance of prototype ultrasound catheters and acoustic protocols. LEF is defined as the percent increase in the rate of lysis compared to tPA alone as measured in our simple clot model consisting of a plasma clot in a static tube. The amount of thrombolysis can be measured by determining the change in fibrin weight compared to control clots. The LEF is expressed as a percent change in the thrombolysis rate compared to lytic control. When the EkoSonic catheter was initially launched, the *in vitro* LEF was about 50% using a constant amplitude pulsed ultrasound protocol with an average power of 2.7 W per transducer group. Later, the MACH4 ultrasound protocol was introduced with an LEF of about 70%. This protocol had a higher average power (3.5W) and consisted of variable pulse amplitudes, which were randomly sequenced. Further investigations with the clot model lead to the development of the currently marketed MACH4e protocol with an LEF of about 90% but with variable amplitude pulses, which cycle up and down. The clot model and LEF are being used to further develop EKOS' technologies.

3:00

2pBA7. Histotripsy-enhanced thrombolysis. Kenneth B. Bader (Radiology, Univ. of Chicago, 231 Albert Sabin Way, CVC 3935, Cincinnati, Ohio 45267-0586, Kenneth.Bader@uc.edu), Samuel A. Hendley (Graduate Program in Medical Phys., Univ. of Chicago, Chicago, IL), Viktor Bollen (Radiology, Univ. of Chicago, Chicago, IL), Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), Kevin J. Haworth, and Christy K. Holland (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Deep vein thrombosis (DVT) is a major public health problem, affecting 600,000 Americans annually with a healthcare cost of $10 billion. Standard interventional techniques are not effective for the chronic thrombus components present in 27% to 43% of DVT cases. Histotripsy is a focused ultrasound therapy that employs the mechanical action of bubble clouds to ablate tissue and induce vigorous fluid mixing. We have demonstrated synergy between histotripsy and the thrombolytic recombinant tissue plasminogen activator (rt-PA) for dissolution of retracted venous clots *in vitro*. Here, the role of histotripsy-induced bubble activity in rt-PA clot dissolution will be

discussed. The rt-PA thrombolytic efficacy of histotripsy pulses that nucleate bubble activity either via (1) shock wave scattering or (2) exceeding the intrinsic cavitation threshold of the clot will be presented. The impact of each histotripsy type on the formed elements that comprise the clot will be described, as well as the likelihood of containing mechanical activity within the target zone. Finally, the utility of passive cavitation imaging to quantify the bubble activity necessary for liquefaction of tissue-mimicking phantoms representative of acute and chronic pathologies will be presented.

Contributed Papers

3:20

2pBA8. Design of a focused ultrasound transducer for histotripsy-thrombolytic combination therapy. Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, 1013 NE 40th St., Seattle, WA 98105, amax38@u.washington.edu), Kevin J. Haworth, Christy K. Holland (Div. of Cardiovascular Health and Disease, Univ. of Cincinnati, Cincinnati, OH), and Kenneth B. Bader (Dept. of Radiology, Univ. of Chicago, Chicago, IL)

Chronic thrombus components are resistant to removal by current interventional techniques and can act as thrombogenic sources. Histotripsy is a focused ultrasound therapy that utilizes the mechanical activity of bubble clouds to liquefy target tissues. *In vitro* experiments have demonstrated histotripsy provides enhancement of the thrombolytic agent recombinant tissue plasminogen activator in a retracted clot model representative of chronic thrombus. Although these *in vitro* results are promising, further refinement of the acoustic source is necessary for *in vivo* studies and clinical translation. To define the source parameters for use *in vivo*, a design study was conducted with transcutaneous exposure of porcine and human iliofemoral deep venous thrombosis (DVT) as the target. Design parameters were selected to confine the focus, and thus bubble activity, within the femoral vein of DVT patients. Furthermore, the therapy array accommodates the placement of a confocal diagnostic linear array for image guidance. Based on the design criteria, a 1.5-MHz elliptical source with 6-cm focal length and a focal gain of 70 was selected. Details of the design, fabrication, and characterization the source will be presented, as well as the means by which bubble activity is initiated.

3:35

2pBA9. Intrinsic threshold versus shock scattering histotripsy treatment of clots *in vitro*. Viktor Bollen (Dept. of Radiology, Univ. of Chicago, 5835 South Cottage Grove Ave., MC 2026, Q301A, Chicago, IL 60637, bollen@uchicago.edu), Kevin J. Haworth (Internal Medicine, Div. of Cardiovascular Diseases and Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH), Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), Christy K. Holland (Internal Medicine, Div. of Cardiovascular Diseases and Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH), and Kenneth B. Bader (Radiology and Committee on Medical Phys., Univ. of Chicago, Chicago, IL)

Catheter-directed thrombolytics are the frontline therapy for iliofemoral deep vein thrombosis. We have demonstrated that histotripsy can enhance the efficacy of the thrombolytic recombinant tissue plasminogen activator (rt-PA) through the mechanical activity of bubble clouds. Depending on the histotripsy pulse duration, bubble activity can be initiated via shock wave scattering or exceeding the peak negative pressure of the intrinsic cavitation threshold. In this study, the thrombolytic efficacy and bubble activity of shock scattering and intrinsic threshold histotripsy were compared. The mass loss was assessed of retracted human whole blood clots exposed to rt-PA (0 or 2.68 μg/mL) and histotripsy pulses of 1 (intrinsic threshold) or 5 (shock scattering) cycles duration. During the histotripsy exposure, bubble cloud emissions were captured on a linear array and processed to form passive acoustic images. The combination of rt-PA and histotripsy was more efficacious than rt-PA alone at 25 MPa peak negative pressure for single-cycle pulses, and 20 MPa peak negative pressure for five-cycle pulses. The degree of thrombolytic efficacy correlated with bubble cloud emissions within the clot. Overall, these results indicate that rt-PA and shock scattering or intrinsic threshold histotripsy is a promising combination thrombolytic therapy.

3:50

2pBA10. *In vitro* assessment of the relationship between medium stiffness and bubble activity for histotripsy-induced liquefaction. Samuel A. Hendley, Gregory Anthony (Committee on Medical Phys., Univ. of Chicago, 5812 S Ellis Ave., IB-016, Chicago, IL 60637, hendley@uchicago.edu), Viktor Bollen, and Kenneth B. Bader (Radiology, Univ. of Chicago, Chicago, IL)

Histotripsy is a focused ultrasound therapy that ablates tissue via the mechanical activity of bubble clouds. While effective for healthy tissues, histotripsy-induced bubble expansion will be mitigated in stiff chronic pathologies. In this study, the bubble cloud activity necessary for liquefaction of agarose phantoms with elastic moduli ranging from 12.3 $\pm$ 3.67 to 142 $\pm$ 44.9 kPa was investigated. Bubble clouds were initiated with 1-MHz pulses of 5-μs duration and peak negative pressures of 12 to 24 MPa. Bubble cloud emissions were mapped via passive cavitation imaging, and correlated with liquefaction using receiver operating characteristic analysis. The maximum power of emissions, and azimuthal location of the maximum power, were recorded for each experimental condition. For phantoms with elastic moduli between 12.3 and 85.8 kPa, no change was indicated in the bubble activity necessary for liquefaction. A larger acoustic power was associated with liquefaction of phantoms with elastic moduli of 142 kPa compared to 22.1 kPa. For a given peak negative pressure of the histotripsy pulse, no change in the peak power or azimuthal location of peak emissions was observed. These results indicate that a fixed bubble activity dose predicts histotripsy liquefaction over a wide range of medium stiffness.

Session 2pED

Education in Acoustics: Acoustics Education Prize Lecture

Benjamin V. Tucker, Cochair
Linguistics, University of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Chair's Introduction—3:25

Invited Paper

3:30

2pED1. The University of New Orleans ocean acoustics program at the Stennis Space Center, Mississippi. Stanley A. Chin-Bing (Dept. of Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, chin-bing@att.net)

In the late 1970's two US Navy ocean R&D organizations were formed and located at the Stennis Space Center (SSC), Mississippi, located approximately 50 miles East of the University of New Orleans (UNO). These two organizations employed nearly 1500 scientists, engineers, and technicians. Many of those with a Bachelor degree in physics, oceanography, or engineering desired advanced training in acoustics and signal processing. In 1982, George E. Ioup took the initiative to have UNO develop and teach acoustic courses on-site at the SSC. In the following 33 years, I developed 11 different graduate level courses in acoustics and taught them multiple times at the SSC. It was possible to take all the necessary courses needed for the Masters and PhD degrees on-site, while maintaining full-time employment. Several dozen Navy scientists received advanced degrees in physics with a specialty in acoustics from UNO. Many more received specific training in acoustics that enhanced their professional careers. This presentation will highlight my contribution to the UNO program at the SSC, and discuss how I incorporated my own research in ocean acoustic propagation and scattering into the many acoustic courses that I taught.

Session 2pID

Interdisciplinary and Education in Acoustics: Promoting Student Publishing Success

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

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

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

Anders Lofqvist, Cochair
Haskins Labs., 300 George St., New Haven, CT 06511

Chair's Introduction—1:00

Invited Papers

1:05

2pID1. Promoting student publishing success: The progressive co-authorship cycle. Ann Bradlow (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, abradlow@northwestern.edu)

I will describe the *"progressive co-authorship cycle"* as a strategy for promoting student publishing success. Crucial points in this progression are: (1) an initial phase in which the advisor is the lead author on a co-authored publication, (2) a middle phase in which the student and advisor are both in transition, and (3) a final phase in which the role reversal is complete and the student is the lead author on a co-authored publication. Each phase serves a unique purpose in the overall progression: modelling of good practice (phase 1), infusion of new energy and dynamism (phase 2), and finally, consolidation of a new and more equitable partnership (phase 3). I will emphasize that, as the fulcrum of the cycle, the middle phase is the most important yet easiest-to-neglect phase in a student-advisor relationship. During this phase, the student develops other co-authorship opportunities (e.g. with fellow students and/or other mentors). By infusing new ideas and research strategies into the mentor-mentee relationship, these "external" collaborations can be particularly enriching for phase 3. While this progression is rarely realized in clearly demarcated phases, it can provide a guiding framework for mentoring, and specifically for supporting student publishing success.

1:25

2pID2. Writing strategies for students and their mentors: From the perspective of a recent student. Kelly L. Whiteford (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, whit1945@umn.edu)

Writing an empirical paper can be a daunting process, particularly for students writing their first manuscript. This presentation will provide advice for helping students develop a structured and goal-oriented writing routine to aid in the publishing process. First, the mentor can help foster motivation to write by guiding the student to pursue research that is tethered to the student's intrinsic interests but still within the domain of the mentor's expertise. Second, writing goals need to be prioritized by the student in a specific, achievable, and time-bound manner. This prioritization can occur through creating a structured, weekly outline of writing-related goals, either with guidance from the mentor or through a peer support group. Setting aside regular, protected writing time can help form a habit out of writing and ensure it does not become pushed aside for other priorities. Lastly, -timely, thorough, and constructive feedback from the mentor can speed up the writing process while conveying to the student that their work is a priority of the mentor.

1:45

2pID3. Empowerment through POMA: How conference proceedings help students publish. Kent L. Gee (Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

When it comes to publishing research results in a thesis or journal article, a disconnect between advisor expectations and student abilities and motivation can be a source of frustration, fear, and deteriorated relationships. Publishing in conference publications, including *Proceedings of Meetings on Acoustics* (POMA), represents an opportunity to bridge this gap. This presentation describes several benefits of publishing in conference proceedings, how to use POMA as a springboard, and some suggestions to help students and advisors overcome the barriers to publication.

2:05

2pID4. Perspectives on mentorship in the world of publishing. Ruth Litovsky (Commun. Sci. & Disord., Univ. of Wisconsin-Madison, 1500 Highland Ave., Waisman Ctr. Rm. 521, Madison, WI 53705, litovsky@waisman.wisc.edu)

Emerging scientists, students and postdoctoral fellows are typically being trained and are developing relationships with their mentors. This presentation will focus on issues that include: (1) timeline and goals related to one's career path; (2) strategic planning in publishing your work during the training period; (3) continuing to publish after moving on to other positions; (4) selecting journals and ethics in determining authorship; (5) how to get a mentor who is quite busy to respond to your drafts in a timely manner; (6) publishing quality vs. quantity; (7) reading and interpreting reviews; (8) getting involved in writing reviews when you advance in your career path. The presenter, Ruth Litovsky, received her PhD in 1991 and is Professor at the University of Wisconsin Madison, where she mentors students, post-docs and junior scientists.

2:25–2:40 Break

2:40–4:10 Panel Discussion

TUESDAY AFTERNOON, 14 MAY 2019 COMBS CHANDLER, 3:00 P.M. TO 5:30 P.M.

Session 2pMU

Musical Acoustics: Bluegrass Music and Related Instruments

Whitney L. Coyle, Chair
Department of Physics, Rollins College, 1000 Holt Ave, Winter Park, FL 32789

Chair's Introduction—3:00

Contributed Papers

3:05

2pMU1. On the phenomenon of natural-frequency splitting of a guitar string caused by a magnetic field. Jack Feinberg (Phys., Univ. of Southern California, Los Angeles, CA 90089-0484, feinberg@usc.du) and Bingen Yang (Aerosp. and Mech. Eng., Univ. of Southern California, Los Angeles, CA)

A magnet can affect the vibration of the metal strings in a musical instrument. We show that the magnetic field from a magnetic pickup can cause a frequency splitting of a metal guitar string's normally degenerate transverse vibration modes, leading to a beat note in the resulting sound. The amount of frequency splitting induced by the magnet depends on the product of the induced magnetization in the ferromagnetic string, and the magnet's spatial gradient at the position of the string, and is on the order of a few Hz. We apply free vibration theory to the string to obtain an eigenvalue problem, which we solve using a distributed-transfer function method. This method accurately predicts the natural frequencies of the vibrating guitar string in a non-uniform magnetic field. Videos of a high-E string vibrating with and without a nearby magnet will be shown.

3:20

2pMU2. The musical saw: Musical acoustics of trapped vibrational modes in a curved blade. Randy Worland (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

The musical saw is a popular folk instrument consisting of a flexible hand saw that is bent into an "*S*"-shape by the performer. When the smooth edge of the saw is bowed, skilled musicians can produce intricate melodies, as well as vibrato and glissando effects, by varying the curvature along the blade. The acoustically significant vibrational modes are trapped near the inflection point between regions of alternating curvature along the saw, exhibiting very low damping as a result of this confinement. Experimental data, taken using electronic speckle-pattern interferometry, illustrate the relevant mode frequencies, shapes, and positions relative to the blade curvature for various configurations of the saw. These data are presented and interpreted along with results from a finite element model of the tapered and curved saw blade.

Invited Papers

3:35

2pMU3. Investigations into the acoustics of the mandolin. Steve Tufte, Samuel Hunt, and Gerrick Hegarty (Dept. of Phys., Lewis & Clark College, MSC 15, 0615 SW Palatine Hill Rd., Portland, OR 97219, tufte@lclark.edu)

The mandolin is a descendent of the lute with a long history starting in 18th century Italy. This talk will give a brief history of the mandolin culminating in the modern flat-back mandolins used in bluegrass music. After a summary of previous research into its acoustics, we will describe our current research. This includes the use of Electronic Speckle Pattern Interferomentry (ESPI) to study the low-modes of the front and back plates and studies of the driving point impedance at the bridge. The mandolin has four courses of doubled strings tuned in unison. Inspired by Weinreich's seminal study of piano strings, we explore the coupling of the strings' motions to each other through their attachment to the vibrating bridge. By using a macro lens on a high-speed camera, we are able to study these motions in exquisite detail. We observe clear evidence of coupling including the exchange of energy between the vertical and horizontal motions of each string and correlations in the phases of their motions. We interpret the experimental results in the context of a theoretical model.
[Work supported by the National Science Foundation under Grant No. 1707978.]

3:55–4:10 Break

Concert
4:10–5:30

Following the technical presentations will be a bluegrass music concert, all are welcome. The local bluegrass group RELIC will perform and discuss the history of the bluegrass genre and the instruments themselves.

Aaron Bibelhauser is a singer, songwriter, and instrumentalist from Louisville, Ky. In addition to writing songs recorded by award winning bluegrass artists including Balsm Range, Del McCoury Band, Michael Cleveland & Flamekeeper, and Dale Ann Bradley, he's taken first place in the Chris Austin Songwriting Contest as Merlefest and earned a nomination for the IBMA's prestigious Song of The Year Award. In addition to being an accomplished solo recording artist, radio broadcaster, and session player, Bibelhauser fronts the Louisville, KY based bluegrass band, Relic, along with his twin brother, Adam. Hailing from a city that bears a rich history of honoring the roots of traditional music while stretching boundaries and embracing progress. Relic is part of revival of this platform, blending rich vocal harmonies with colorful instrumentation.

Session 2pNS

Noise, Architectural Acoustics, ASA Committee on Standards, and Animal Bioacoustics: Soundscape and its Application Based on the New Standard

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

Klaus Genuit, Cochair
HEAD acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square - Suite 103, Vernon, CT 06066

Chair's Introduction—1:30

Invited Papers

1:35

2pNS1. The soundscape standard—Its development and challenges. Brigitte Schulte-Fortkamp (TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

Soundscape entered the debate of noise annoyance issues in the nineties, most likely at the same time as when the EU Directive 2002/49/EC [PJ1] on Noise was under development. In 2014, the ISO standard ISO 12913-1: "Acoustics-Soundscape-Part 1: Definition and conceptual framework" was published, delivering the first framework for soundscape. This was followed in 2018 by "ISO/TS 12913-2 Acoustics—Soundscape—Part 2: Data collection and reporting requirements", which provided evaluation processes and information for integrating stakeholder participation. The third part of the standard is now underway: ISO/ TS 12913-3 "Acoustics- Soundscape—Part 3: Data analysis." Together, the international standard ISO 12913 provides guidance for accessing the key components in soundscape: people, acoustic environment, and context. The available standards in soundscape, psychoacoustics and noise management have provided a large step towards enhancing the quality of life for people. the urgent need for recognition but also for application with regard to new understandings of urbanism will be discussed.

1:55

2pNS2. Consequences in standardizing the soundscape data collection process. André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de)

The ISO/TS 12913-2 dealing with data collection and reporting issues in the context of soundscape investigations was published in 2018. This technical specification proposes a range of methods and tools to perform the data collection process according to the current understanding of the soundscape approach. It intends to harmonize the data collection of soundscape studies and to ensure a certain level of comparability between soundscape investigations. In the past the vast majority of soundscape studies applied very different methods and tools limiting the comparability of results and avoiding the performance of meta-analyses. The reason might be the interdisciplinary roots of the soundscape concept asking for diverse disciplinary accesses to the phenomenon challenging the soundscape standardization efforts. Now, the disadvantage of the limited comparability of soundscape results might be overcome due to the publication of the ISO/ TS 12913-2. The paper will discuss the expected impact of the technical specification defining the data collection requirements on soundscape research and will highlight the benefits and potential drawbacks in standardizing the holistic soundscape approach.

2:15

2pNS3. Binaural measurement and psychoacoustic analysis—An advantage for the environmental noise research. Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de)

The new ISO TS 12913-2 "Data collection and reporting requirements" describes a lot of new aspects how to analyze soundscape (e.g., environmental noise) and provides new tools for the measurement and analysis. Whereas the use of binaural recording and psychoacoustic analysis is well known in the field of product sound quality especially with respect to the automotive field, the application of these tools within the soundscape analysis is new. This paper illustrates based on different examples given from the field of environmental noise the superiority and advantage of these tools in comparison to the conventional measurement using an omnidirectional microphone calculating the A-weighted sound pressure level.

2:35

2pNS4. An experimental soundscape study, combining binaural recordings, *in situ* questionnaires and behavioral mapping. Antonella Radicchi (Institut für Stadt- und Regionalplanung, Technische Universität Berlin, Hardenbergstraße 40 a, Sekr. B 4, Berlin, Berlin 10623, Germany, antonella.radicchi@tu-berlin.de)

In 2014 and, more recently, in 2018 the ISO norms on soundscape were released with the aim of providing a conceptual framework and standardized data collection methods for the international community of scientists and professionals with an interest on soundscape. This paper presents a research study conducted in the summer of 2018 in a Berlin public square where, following the ISO norms on soundscape, methods were applied to further investigate the findings of the "Beyond the Noise: Open Source Soundscapes" project, i.e., the everyday quiet areas as identified by the participants in the project. First, the paper introduces the research questions originated from the previous study. Second, it outlines the fieldwork procedure and methods, consisting in binaural measurements made in parallel to *in situ* questionnaires and behavioral mapping. Third, it discusses the initial results and limitations of the study. In conclusion it argues the soundscape approach potential within urban design and planning, illustrating how the results were partially exploited to design a bottom-up master plan proposal so as to improve the public square's soundscape and reduce noise pollution.

2:55–3:10 Break

3:10

2pNS5. Soundscape for smart tourism in Macao. W M To (Macao Polytechnic Inst., Macao, Macao) and Andy Chung (Macau Instituto de Acoústica, Macao, Macao, Macao, ac@smartcitymaker.com)

This paper presents a new initiative in enhancing the experience of tourists visiting Macao, by way of appling soundscape based on the new technical specification ISO/TS 12913-2:2018.

3:30

2pNS6. Prediction model of soundscape assessment based on semantic and acoustic/psychoacoustic factors. Ming Yang (HEAD Acoust. GmbH, Herzogenrath 52134, Germany, mingkateyang@163.com)

While conventional sound/noise environment assessment, management and relevant standards are based on the objective measurement of sound pressure level (SPL), soundscape research showed that humans' subjective assessment (such as positive and negative) cannot be well explained by the SPL or other acoustic indicators alone. Rather, soundscape assessment depends on many more factors such as cognition and context, which also need to be considered as suggested in the new soundscape standard. Among the different factors, a large number of previous studies supported that semantic factors, i.e. the sound sources which compose the sound environment—e.g., natural sounds (moving water, bird song, etc.), mechanical sounds (road traffic, construction, etc.) and human activity sounds, play the most important role in human perception and assessment. Therefore, from the systematic literature review of the relationship between such semantic factors and soundscape assessment, this present study proposes a model based on the sound source information, as well as SPL and psychoacoustic indicators of each of the different sound sources, to explain/predict largely, even though not fully, the soundscape assessment. Furthermore, for the practical use of this model which would be independent of manual efforts in recognition of sound sources through listening, this study discusses the possibility of integrating a computer algorithm of sound source recognition from field recordings of sound environment, for automated soundscape assessment.

3:50

2pNS7. Automated parsing of sound level meter data into artifacts, natural sources, and noise. Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

The National Park Service (NPS) has nearly one million hours of 1/3rd octave band sound level measurements collected at hundreds of sites throughout the system. One of the principal purposes of these data is to measure the background or residual sound level against which all transient sounds are heard. Historically this analysis has depended upon trained listeners to identify which sound sources can be heard in segments of these data. These annotated records were then used to calculate an adjusted median sound level. To pursue a more efficient, automated method to deliver this result, matrix decomposition methods were tested to characterize their capacity to model time series of sound level spectra as low-rank decompositions, including options to account for anomalous events. These methods were combined with sound source identification based on component spectral properties and time weightings to yield promising approaches for automating NPS analyses. These results may also offer options for detecting and removing the effects of pseudonoise due to the turbulence generated by air flowing past the microphone.

Contributed Paper

4:10

2pNS8. Evaluating manufacturing environment soundscapes. Brian J. Puckett and Erica E. Ryherd (Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S 67th St., Omaha, NE 68182, puckett.brian@huskers.unl.edu)

This work presents an investigation into the soundscape of a large, industrial manufacturing plant in Lincoln, Nebraska, including exploration of the noise reduction performance of sound enclosures. Occupational noise-induced hearing loss is among the most common nonfatal work-related injuries and illnesses in the U.S., and most efforts to reduce this exist as limited noise exposure times and mandatory use of hearing protection devices. This research aims to develop knowledge of factory soundscapes and explore acoustical strategies to reduce noise near the source. The factory in this study uses two distinct types of sound enclosures on site: rigid box-style enclosures and resilient curtain-style enclosures. Using state-of-the-art technology, it was possible to better understand contributions of single noise sources to the complex, overall soundscape and evaluate some detailed acoustical characteristics of the noise. The measured data are compared with subjective assessment of company employees using methods described in Part 2 of the Soundscape Standard (ISO/TS 12913-2). Results presented include a noise reduction comparison of the two types of enclosures, a relationship between noise reduction and source-enclosure proximity, and a comparison between measurements and survey data.

4:25–4:45 Panel Discussion

Session 2pPA

Physical Acoustics and Noise: Nonlinear Acoustics for Non-Specialists II

Won-Suk Ohm, Cochair
Yonsei University, 50 Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea

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

Chair's Introduction—1:30

Invited Papers

1:35

2pPA1. Curious nonlinearity of rocks. Carly M. Donahue and Paul A. Johnson (Earth and Environ. Sci., Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, cmd@lanl.gov)

Many geological materials, ranging from "rocks to unconsolidated sand," exhibit highly nonlinear elastic properties. Rocks fall in to a class of materials know as Nonlinear Mesoscopic Elastic Mesoscopic materials (NMEMs) in which the nonlinearity they possess is not derived from the constituent material, but rather the microscopic structure. Their behavior manifests as characteristic wave distortion, and slow dynamics, a recovery process to equilibrium that takes place over hours, days, weeks and sometimes years after a wave disturbance. A number of acoustic techniques have been developed to quantify a the material's nonlinear elastic coefficients and image localized damaged areas; and while much has been learned, much is left unknown, particularly identifying and understanding the underlying physical mechanisms that give rise to a rock's nonlinear elastic response, such as frictional losses, soft regions, and the influence of water content. But rocks also are a platform to understand other localized and distributed nonlinearity. Nondestructive evaluation techniques of metal and concrete are being developed with applications towards crack detection and wellbore integrity. Nonlinear acoustic techniques have recently appeared promising for performance evaluation of pressed powders and additively manufactured materials. In this presentation, we will provide an overview of nonlinear elasticity as illustrated by some examples.

1:55

2pPA2. Nonlinear acoustic metamaterials. Samuel P. Wallen (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Mark F. Hamilton, and Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu)

Acoustic metamaterials (AMM) have become a very active topic for research in numerous domains of engineering and science because of their promise to create materials, structures, and devices that can control acoustic wave propagation in ways that exceed the capabilities of naturally occurring or conventional composite materials. The majority of AMM research has been focused on linear behavior such as negative dynamic effective stiffness and density, cloaking, and negative refraction. One drawback of the focus on linear behavior is the restriction of the effective material properties of interest to narrow frequency bands that cannot be changed with external stimulus. Nonlinearity has been explored as a means of increasing the bandwidth of performance by enabling tunable band gaps and material configurability. Other recent work has investigated nonlinear AMM to access phenomena such as harmonic generation, non-reciprocity, enhanced energy absorption, solitons, mode hopping and conversion, chaos, and intrinsic localized modes. This talk will provide an overview of nonlinear AMM, starting with a background on AMM, then surveying existing research on the topic and its relationship to seminal works in nonlinear acoustics, and finally discussing promising avenues of future research. [Work supported by the National Science Foundation EFRI program and ARL:UT.]

2:15

2pPA3. Nonlinear acoustics of complex solids and granular media. Vincent Tournat (LAUM, UMR CNRS 6613, Le Mans Université, Av. O. Messiaen, Av. O. Messiaen, Le Mans 72085, France, vincent.tournat@univ-lemans.fr)

This talk is an introduction to nonlinear acoustic processes in complex solids such as cracked metals, damaged composites or concrete, and granular media. These materials, belong to the same class of complex solids often referred to as mesoscopic solids, and share a number of similar nonlinear acoustic behaviors originating mainly from the presence of internal solid contacts. The specific nonlinear acoustic signatures and processes will be described (slow dynamics, memory, contact acoustic nonlinearity, i.e. nonclassical nonlinearities) through several examples, and an overview of the main modeling tools will be provided. Finally, the current trends and several applications of such nonclassical nonlinear processes (nondestructive testing and characterization, wave control, etc.) will be given.

2:35

2pPA4. Nonlinearity in high amplitude focused waves in solids for crack detection using time reversal techniques. Brian E. Anderson, Sarah M. Young (Dept. of Phys. & Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, bea@byu.edu), Marcel Remillieux (Geophys. Group, Los Alamos National Lab., Los Alamos, NM), Pierre-Yves Le Bas, and Timothy J. Ulrich (Detonation Sci. and Technol. Group, Los Alamos National Lab., Los Alamos, NM)

Defects in a sample cause localized distortions of a wave. Focused waves at a defect create amplitude-dependent signatures that may be quantified in post processing, thereby offering a means of locating and characterizing the defect. Time reversal (TR) is a technique that allows intentional focusing of wave energy. TR has been developed for crack and defect detection in solid media. Because TR provides a localized focus of elastic energy, it can be used to study the local nonlinear properties at point(s) of interest that are indicative of the presence of cracks and defects. Source transducers may remain in place as a sample is scanned for defects. A laser Doppler vibrometer offers a noncontact means of selecting points at which to focus energy. TR applied to detect cracking in steel rods will be presented. The results demonstrate that cracks may be identified through their nonlinear signatures when TR is used to focus energy at various positions along the rods.

2:55

2pPA5. On the utility of C/S as an indicator of nonlinearity in finite-amplitude dispersive waves. Taeyoung Park, Won-Suk Ohm (Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, ohm@yonsei.ac.kr), Kent L. Gee, and Brent O. Reichman (Brigham Young Univ., Provo, UT)

A frequency-domain nonlinearity indicator *C/S*, recently introduced by Ohm *et al.* [*Proc. Mtgs. Acoust.* **29**, 045003 (2016)], is a companion indicator to the well-known Morfey-Howell *Q/S* [*AIAA J.* **19**, 986–992 (1981)]. The two indicators can be regarded as "two sides of the same coin," because *C/S* and *Q/S* are, respectively, defined by the real and imaginary parts of the cross-spectrum of the pressure and squared-pressure waveforms. Despite the common origin, however, *C/S* and *Q/S* describe the pertinent nonlinear wave process in a different light. Furthermore, for nonlinear waves in media with strong dispersion only *C/S* may be applicable, because the use of *Q/S* could lead to erroneous interpretations. In this paper, we demonstrate the utility of *C/S* in the context of the Korteweg-de Vries (KdV) equation, which is one of the most celebrated model equations for nonlinear waves in dispersive media. The success of *C/S* is juxtaposed with the limitation of *Q/S* in describing nonlinear dispersive wave processes, especially in the case of solitons.

3:15–3:30 Break

Contributed Papers

3:30

2pPA6. Laboratory study of nonlinear propagation of transient signals in a sandstone bar. Thomas G. Muir, John M. Cormack, Charles M. Slack, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, P/O. Box 8029, Austin, TX 78713, muir@arlut.utexas.edu)

Experiments are reported on finite-amplitude, impact propagation in a thin bar of Texas moss sandstone. Wideband, unipolar pulses were generated by excitation with a pendulum hammer at one end of the bar, in contrast to prior work with either tone-burst excitation or resonance methods. The bar was rectangular, 8 cm on each side and 175 cm in length, with a density of 2.0 g/cc. Impacts had a center frequency around 3 kHz, amplitude of 10 to 130 microstrain, and propagated repetitively between reflections at each end. Measurements were made with a laser Doppler vibrometer at multiple locations along the bar, while ultrasonic tone-burst probes were used at a central location, transverse to the axis. The sound speed and effective modulus were reduced in proportion to the impact amplitude as the finite-amplitude impact pulse propagated through the transverse and other measurement sites. The attenuation of the impact signal was also found to follow a power law in the frequency range of 1–6 kHz, in proportion to impact intensity. Although amplitude-dependent attenuation and softening of the elastic modulus are both found in micro-inhomogeneous materials, the present results were enhanced, up to an order of magnitude, compared to those in previously reported experiments.

3:45

2pPA7. Modeling of plane progressive waves in media that exhibit slow dynamics and dissipative nonlinearity. John M. Cormack, Thomas G. Muir, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

Nonclassical nonlinearity refers to amplitude-dependent effects that are irreversible, for example slow dynamics and dissipative nonlinearity, both phenomena that are commonly observed in materials such as sandstone and concrete. Analyses of progressive waves in such media have been few, and have relied on hysteresis models that are difficult to implement, especially for transient or strongly nonlinear pulses. In this work, an evolution equation for plane progressive waves in a nonclassically nonlinear material is derived using two phenomenological models: the internal variable model of Berjamin *et al.* for slow dynamics [Proc. Roy. Soc. A **473**, 20170024 (2017)], and a simple model for dissipative nonlinearity [Zaitsev and Nazarov, *Acoust. Phys.* **44**, 362 (1998)]. The evolution equation is easily solved numerically given an arbitrary initial waveform. An initially narrowband signal exhibits pulse lengthening, amplitude-dependent attenuation, and nonclassical waveform steepening during propagation. Experimental observation of the same phenomena were reported recently by Remillieux *et al.* [*J. Geophys. Res.* **122**, 8892 (2017)]. Their results are reproduced in this study using the new evolution equation, showing good agreement between the model and the measurements, with additional insight into the behavior of the material gained from the model. [J.M.C. supported by the ARL:UT McKinney Fellowship in Acoustics.]

4:00

2pPA8. Direct detection of dark matter particles by nonlinear effects in crystal vibrations. Igor Ostrovskii (Phys. and Astronomy, Univ. of Mississippi, Lewis Hall, Rm. 108, University, MS 38677, iostrov@phy.olemiss.edu)

The energetic dark matter particles (EDP) of $(4+1.5) \times 10^{15}$ eV mass-energy propagating from the central region of the Milky Way galaxy (CMW) are directly detected with the help of half-inch ~10 MHz quartz vibrators. The spectra of acousto-electric vibrations of thin piezoelectric crystal are theoretically calculated, and a nonlinear effect of external frictional influence is taken into account. The EDP produce the changes in the spectra of vibrations under specific orientation of the crystallographic axes toward the CMW. The mechanism of EDP-crystal interaction may be associated with the gravitational influence of EDP and their quantum properties, such as matter waves. The EDP average energy density is $0.71+0.29$ (GeV/cm^3) near Earth, which agrees with the expectations and existing astrophysical bounds of dark matter density in the solar neighborhood. The EDP can

transfer their momenta-energy to a media of propagation. The influences of EDP, or naviten, on vibrating atoms are detected from 10 different samples during 5 years. The micro-void tracks of EDP-particles are also observed in the vibrating plates of fuzzed quartz, which appear due to EDP decay inside solid media. The existence of EDP-particles is in agreement with recent detection of the ~10^{15} eV energy neutrinos from the CMW region.

4:15

2pPA9. Evolution equation for nonlinear Lucassen waves. Blake E. Simon, John M. Cormack, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, blakesimon8@utexas.edu)

A nonlinear, fractional, surface-wave equation was developed recently by Kappler *et al.* [*Phys. Rev. Fluids* **2**, 114804 (2017)] for propagation along an elastic interface coupled to a viscous incompressible medium. Linear theory for attenuation and dispersion of such a wave was developed originally by Lucassen [*Trans. Faraday Soc.* **64**, 2221 (1968)]. Kappler *et al.* employ a fractional derivative to account for the Lucassen attenuation and dispersion, and they include quadratic and cubic nonlinearity of the elastic interface. Presented here is a simplified form of their model equation for plane progressive waves. The resulting nonlinear evolution equation has the form of a Burgers equation but with a fractional derivative in place of the second derivative for viscosity, and with cubic as well as quadratic nonlinearity. In addition to facilitating analytical and numerical calculations, the evolution equation enables interpretation of a threshold phenomenon, revealed in numerical simulations presented by Kappler *et al.*, as competition between quadratic and cubic nonlinearity. It is also suitable for determining critical source amplitudes above which Lucassen attenuation and dispersion alone cannot preclude formation of unphysical multivalued waveforms [Cormack and Hamilton, *Wave Motion* **85**, 18 (2019)]. [BES and JMC were supported by the ARL:UT McKinney Fellowship in Acoustics.]

4:30

2pPA10. Nonlinear acoustic crack detection in thermoelectric wafers. John Greenhall, Alan Graham, and Cristian Pantea (Mater. Phys. and Applications, Los Alamos National Lab., MS D429, Los Alamos, NM 87544, jgreenhall@lanl.gov)

We present an acoustic technique for noninvasive crack detection in small (approximately 30.0 $\times$18.0 $\times$ 0.5 mm) thermoelectric wafers. The technique is based on exciting the wafers with a low-frequency signal that drives the crack to open and close periodically, and a high-frequency signal that is permitted to propagate through the closed crack and prohibited from propagating through the open crack. Interaction between the low- and high-frequency signals and the crack leads to generation of acoustic nonlinearities in the wafer. In contrast to existing acoustic crack detection techniques we utilize standing waves within the wafers to facilitate simultaneous crack detection throughout the wafer, we do not require uniform dimensions and material properties between wafers, and we do not affix the transducer to the wafers to avoid damaging the wafers. We present a mathematical model of the acoustic nonlinearity generation process and develop a procedure for identifying cracked wafers. We implement this technique experimentally and correctly identify cracked and crack-free wafers with total error in low single digits. This acoustic crack detection technique finds application in manufacturing of thermoelectric wafer and other wafers, where identifying cracks early in the manufacturing process results in significant time and cost savings.

4:45

2pPA11. Slow-dynamical nonlinearity in a glass bead pack probed with diffuse ultrasound and coda-wave interferometry. Richard Weaver, John Y. Yoritomo (Dept. of Phys., Univ. of Illinois, 1110 West Green St., Urbana, IL 61801-3080, yoritom2@illinois.edu), John Popovics, and James Bittner (Dept. of Civil and Environ. Eng., Univ. of Illinois, Urbana, IL)

Low amplitude diffuse coda ultrasound probe signals received through unconsolidated glass bead packs under static vertical loads are used to measure the bead pack's slow dynamics, in which the speed of low amplitude probe waves varies slowly with time long after loading. Coda-wave interferometry reveals tiny stretches in the waveform, with precision of parts in 10^5. These stretches are ascribed to decreased contact stiffnesses or decreased sound speeds. The coda-wave stretch is observed to be sensitive to low frequency conditionings at strains of order 10^{-5}. Such conditionings include taps on the sides, low frequency harmonic vibrations and impulses on top, and step changes in vertical load. Typical behavior after the conditioning is an initial loss of stiffness followed by a slow healing that proceeds with the logarithm of time. The initial loss of stiffness and its slow dynamic healing are investigated for sensitivity to sundry parameters.

Session 2pPPa

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Elizabeth A. Strickland, Chair
Speech, Language, and Hearing Sciences, Purdue University, 500 Oval Drive, West Lafayette, IN 47907

Chair's Introduction—1:00

Invited Paper

1:05

2pPPa1. Differences and similarities of peripheral auditory systems. Glenis R. Long (Speech-Language-Hearing Sci., Graduate Ctr. CUNY, 365 Fifth Ave., New York, NY 10016, glong@gc.cuny.edu)

The properties of the ears of the wide range of species processing auditory signals can vary greatly depending on both important environmental sounds, and the sounds generated by individual members of the species (which are shaped by the sound processing and sound generation systems). Unfortunately, most anatomical and physiological research is conducted on a small subset of mammals and there are many pressures on auditory researchers to limit their research to species that are easy to work with in laboratories which are motivated to explain normal and impaired hearing in humans. Psychoacoustics has played a major role in our understanding of auditory processing in humans, but is difficult and time consuming in nonhuman species. Much of the investigation in nonhuman species is invasive and not appropriate for use with humans. Otoacoustic Emissions (sounds generated by the inner ear and measured in the outer or middle ears) can be used in most species, but we are still investigating the extent to which OAE obtained from different species reflect similar underlying processes. The combination of OAE measurement and computer modelling of the data has made us aware of the importance of evaluation of cross-species and individual differences in auditory processing.

Session 2pPPb

Psychological and Physiological Acoustics and Education in Acoustics: Cultivating New Growth by Composting Old Ideas: Pruning the Deadwood from the Garden of Psychological and Physiological Acoustics

G. Christopher Stecker, Chair
Hearing and Speech Sciences, Vanderbilt University, 1215 21st Ave. South, Room 8310, Nashville, TN 37232

Chair's Introduction—2:10

Invited Papers

2:15

2pPPb1. Critical examination of critical bands. Laurel H. Carney (Biomedical Eng. & Neurosci., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu)

The concept of the critical band has driven the design and interpretation of psychophysical experiments for decades. The critical band was initially proposed by Fletcher [*Rev. Mod. Phys.* **12**, 47 (1940)] to describe results of masking experiments, and it is generally interpreted as evidence for a filter that limits the signal-to-noise ratio in stimulus representations that are provided to the central nervous system. This filter-based interpretation of masked-detection thresholds has been extended and applied to numerous tasks, including many without masking. The concept of a bank of critical-band filters also forms the basis for signal-processing strategies used for hearing aids and cochlear implants. This talk will review both psychophysical and physiological results that challenge the concept of the critical-band filter. For example, the minimal effect of roving-level paradigms on masked-detection thresholds directly refutes the critical-band based power-spectrum model. Furthermore, most auditory neurons have receptive fields that are much broader than critical bands at the moderate to high sound levels used in most psychophysical tasks, with or without masking. Alternative concepts that are robust across a wide range of levels and in roving-level paradigms, such as fluctuation profiles, explain the original masked-detection results and related experiments, such as psychophysical tuning curves.

2:40

2pPPb2. Challenging standard practices in adaptive psychophysics. Eric C. Hoover (Dept. of Hearing and Speech Sci., Univ. of Maryland, 7251 Preinkert Dr., College Park, MD 20742, ehoover@umd.edu), Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., Portland, OR), and David A. Eddins (Dept. of Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

One of the most cited manuscripts in behavioral hearing research is Levitt [*J. Acoust. Soc. Am.* **49**, 467–477. (1971)]; it is typically the only reference cited in the procedures section of studies reporting thresholds obtained via adaptive tracking. This is problematic because Levitt (1971) informs only one among many parameters used in threshold estimation, and the relationship between Levitt's transformed up-down method and the proportion correct targeted by the procedure is commonly disrupted by the selection of the remaining parameters [*Vis. Res.* **38**, 1861–1881. (1998)]. We explored the evidence supporting standard practices and found that step size, stopping criterion, and threshold estimation from track data have limited theoretical motivation and can introduce systematic bias and error into threshold estimates as shown in behavioral and simulation studies. As a result, we propose an approach to developing and describing methods for behavioral threshold estimation that offer improved efficiency, reliability, reproducibility, and are no less consistent with Levitt (1971). [Work supported by NIH NIDCD DC015051.]

3:05

2pPPb3. Accounting for task switching when measuring listening effort: A cautionary tale for the dual-task paradigm. Adrian K. C. Lee (Speech and Hearing Sci., Univ. of Washington, Box 357988, Seattle, WA 98195, akclee@uw.edu)

Listening in everyday environments inevitably involves task switching. For example, one might be typing up an abstract while listening to children singing in the background or one might be listening to a radio news program while finding a parking space on a crowded street. While there is an extensive body of research that examines how cognitive control processes support the ability to adjust behavior dynamically, how task-switching impacts listening is less explored. In this talk, different kinds of costs associated with task switching will be reviewed. Specifically, how these well-studied switch costs could impact our interpretations of dual-task paradigm findings—an experimental framework that is often used to interrogate listening effort—will also be discussed. [Work supported by NIH R01 DC013260].

3:30

2pPPb4. Circling back on theories of sound localization. Antje Ihlefeld, Nima Alamatsaz (Biomedical Eng., New Jersey Inst. of Technol., 323 Martin Luther King Blvd., Fenster Hall, Rm. 645, Newark, NJ 07102, ihlefeld@njit.edu), and Robert M. Shapley (Ctr. for Neural Sci., New York Univ., New York, NY)

An important question of human perception is how we localize target objects in space. Through our eyes and skin, activation patterns on the sensory epithelium suffice to cue us about a target's location. However, for our ears, the brain has to compute where a sound source is located. One important cue for computing sound direction is the time difference in arrival of acoustic energy reaching each ear, the interaural time difference (ITD). With behavioral experiments on sound lateralization as a function of sound intensity, we tested how the computation of sound location with ITDs is done. We tested twelve naïve normal-hearing listeners (ages 18–27, five females). Stimuli consisted of low-frequency noise tokens that were bandlimited from 300 to 122 Hz, from 5 to 25 dB sensation level. Without response feedback, listeners were initially trained to reliably judge the direction of a sound source and then tested on where they heard the sound. We found that softer sounds tend to be localized closer to midline as compared to louder sound. This finding raises doubt on one major theory of sound localization, the labeled-line theory, and supports another main contender, population rate based coding.

3:55–4:10 Break

4:10

2pPPb5. "Straightness" versus "briefness" in binaural cue extraction. G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu), Mathias Dietz (Universität Oldenburg, Oldenburg, Germany), and Richard M. Stern (Carnegie Mellon Univ., Pittsburgh, PA)

The neural mechanisms that detect and encode interaural delays have elicited intense interest for many decades, from Jeffress's [*J. Comput. Physiol. Psychol.* **41**, 35–39 (1948)] original "place theory of sound localization," its modeling via frequency-specific cross-correlation starting with Colburn [*JASA* **54**, 1458–1470 (1973)], and continuing to this day. An important advance was the emphasis of cross-correlation peaks that align across frequency ["straightness" ; Stern *et al.*, *JASA* **84**, 156–165 (1988)]. While narrowband cross-correlation produces multiple peaks and is therefore inherently ambiguous, straightness detection accounts for resolution of that ambiguity in broadband stimuli by combining binaural outputs across frequency. An alternative account is suggested by the effects of temporal fluctuations, such as onsets, on the *inputs* to binaural processing. Zurek [*JASA* **67**, 953–964 (1980)] described how emphasizing such events (e.g., by "windowing" the input) also resolves interaural ambiguities. Recent psychophysical and physiological evidence supports that view, strongly suggesting that binaural processing does, in fact, occur in brief windows triggered by envelope fluctuations such as onsets and intrinsic fluctuations in bands of noise. This talk investigates the possibility that the resulting temporal sparsity of binaural inputs ("briefness") might account for bandwidth effects even within single frequency channels, i.e. without a second level of "straightness" detection.

4:35

2pPPb6. What studies of audio-visual integration do not teach us about audio-visual integration. Ross K. Maddox (Biomedical Eng. and Neurosci., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rm. 5.7425, Rochester, NY 14642, ross.maddox@rochester.edu)

Auditory perception depends on more than just the processing of acoustic stimuli. Visual stimuli can also have a profound influence on listening. Salient examples of such effects include spatial ventriloquism—in which the location percept of an auditory stimulus is "captured" by that of a simultaneous visual stimulus—as well as drastically improved understanding of speech in noise when the talker's face is visible to the listener. These phenomena are typically described as "audio-visual integration," and are often well modeled, as in the case of the ventriloquist effect, by ideal Bayesian causal inference. However, there is an over-reliance in these studies on single pairs of stimuli (i.e., one auditory and one visual stimulus) and the nature in which cross-modal discrepancies are resolved. This talk will first discuss two problems resulting from that: first, there is ambiguity about whether the integration occurs from weighing two independent sensory estimates or a single bound percept, and second, the design is less useful for studying integration when the stimuli are congruent. The talk will then describe recent work from our lab focused on new designs using multiple stimuli in an attempt to alleviate these issues and inform better models of integration.

Contributed Papers

5:00

2pPPb7. Revisiting perceived intracranial lateralization for stimuli with interaural time differences that are larger than the head. Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, 0119E Lefrak Hall, College Park, MD 20742, goupell@umd.edu), Virginia Best (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA), Julia Nothaft, and H. Steven Colburn (Dept. of Biomedical Eng., Boston Univ., Boston, MA)

The interaural time difference (ITD) is the primary sound-localization cue for humans. While realistic sound sources have energy across a wide frequency range, the ear performs a narrowband frequency decomposition, and within each band are ambiguities in the interaural cross-correlation (an index of the signal ITD). These ambiguities are thought to be resolved by the approximate consistency of ITDs across frequency, or "straightness." However, straightness has not been evaluated over a wide range of stimuli. Therefore, normal-hearing listeners reported the intracranial lateralization of narrowband noises and tone complexes to better evaluate across-frequency ITD processing. In a new modification of the typical paradigm, listeners were encouraged to give multiple responses if split images were perceived. ITDs larger than those naturally produced by the head (~750 ms) best demonstrate straightness because across-frequency comparison is necessary to resolve the interaural phase ambiguities, which is why ITDs as large as 1500 ms were applied. Straightness effects reported previously for narrowband noises were replicated, and were broadly similar for complex tones. The extent of lateralization of 1500-ms ITDs and the occurrence of split images were difficult to account for using simple lateralization-based models for smaller ITDs.

5:15

2pPPb8. Flaws in the use of spectral ripples in cochlear implants. Matthew Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, 1417 NE 42nd St., Seattle, Washington 98105, mwinn83@gmail.com) and Gabrielle O'Brien (Speech & Hearing Sci., Univ. of Washington, Seattle, WA)

Spectral ripple discrimination is a popular measure of spectral resolution that has been shown to correlate with speech recognition scores in cochlear implant (CI) listeners. In the test, listeners distinguish sounds with varying density of spectral peaks, with some spectral modulation depth. We argue that there are numerous significant flaws with the application of the test specifically in CI listeners. To start, the spectrum is aliased by the CI processor in a way that is similar to frequency aliasing for under-sampled time series. Beyond a critical spectral density, the spectral envelope changes in a chaotic fashion and is no longer under experimenter control. This critical density is exceeded in numerous published studies. Furthermore, the densities linked with "good" performance are not only outliers, but are entirely unrelated to the spectral densities of real speech sounds, and likely exhibit undue leverage over correlation values. Additionally, there are reports of experience and learning effects, inconsistent with the often-stated goals of the test to avoid such factors. We show how artefactual nonlinearities at high spectral densities may unintentionally match the spectral envelope characteristics of speech sounds—an unfortunate result that likely has given spurious results that sustain the use of this test.

TUESDAY AFTERNOON, 14 MAY 2019 STOPHER, 1:15 P.M. TO 4:40 P.M.

Session 2pSA

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

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

Alexey S. Titovich, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd, West Bethesda, MD 20817

Invited Papers

1:15

2pSA1. Plate mechanical metamaterials and their applications. Igor Bargatin (Mech. Eng. and Appl. Mech., Univ. of Pennsylvania, 220 South 33rd St., Philadelphia, PA 19104, bargatin@gmail.com)

Recently, we introduced the concept of *plate mechanical metamaterials*—cellular plates with carefully controlled periodic geometry and unique mechanical properties—as well as its initial realization in the form of freestanding corrugated plates made out of an ultrathin film. We used atomic layer deposition (ALD) and microfabrication techniques to make robust plates out of a single continuous ALD layer with cm-scale lateral dimensions and thicknesses between 25 and 100 nm, creating the thinnest freestanding plates that can be picked up by hand. We also fabricated and characterized *nanocardboard*—plate metamaterials made from multiple layers of nanoscale thickness, whose geometry and properties are reminiscent of honeycomb sandwich plates or corrugated paper cardboard. Ultralow weight, mechanical robustness, thermal insulation, as well as chemical and thermal stability of alumina make plate metamaterials attractive for numerous applications, including structural elements in flying microrobots and interstellar light sails, high-temperature thermal insulation in energy converters, photophoretic levitation, as well as ultrathin sensors and resonators. I will briefly discuss our experimental progress on all these applications, including demonstrations of extremely robust thermal insulators that can sustain a temperature difference of ~1000 °C across a micron-scale gap, macroscopic plates that levitate when illuminated by light, and hollow AFM cantilevers that offer greatly enhanced sensitivity and data acquisition rates.

1:35

2pSA2. Effects of geometry and mass distribution in 3D printed metastructures for vibration mitigation. Ignacio Arretche, Ganesh U. Patil, and Kathryn H. Matlack (Dept. of Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, 1206 W. Green St., Urbana, IL 61801, kmatlack@illinois.edu)

This presentation will discuss our recent progress on mechanical metastructures, which have engineered micro- and meso-scale features and are designed to have prescribed vibration mitigation properties. The goal of this work is to understand how the geometry and structure of a material can manipulate elastic wave propagation. Specifically, we developed a class of 3D printed metastructures that support low and wide band gaps, by combining periodic lattices with embedded resonant inclusions. The band gaps in these metastructures have significant shifts in frequency with small geometry changes in the unit cell design. We use finite element method simulations to calculate how band gaps in infinitely periodic metastructures depend on how the material is distributed throughout the unit cell. We study different lattice geometries that exhibit a variety of bend- and stretch-dominated deformations, as well as different volume fraction and

placement of resonant inclusions. We fabricate metastructures with 3D printing and measure their frequency-dependent vibration transmission to experimentally validate their behavior. These metastructures have engineering applications in structural components that prevent the propagation of damaging structural vibrations.

Contributed Papers

1:55

2pSA3. Randomness in periodic metamaterials. Hussein Esfahlani and Andrea Alu (Photonics Initiative, Adv. Sci. Res. Ctr., 85 St. Nicholas Terrace, The Towers at CCNY, Rm. 1G16, New York, NY 10031, hsesfahlan@gmail.com)

The most straightforward way to design artificial structures is to use periodicity and leverage its potential mathematical and physical consequences to our favor. However, strict periodicity, which is usually employed in designing artificial materials, is not practically feasible and most often some degrees of randomness are introduced in the fabrication process, which degrade the performance. In this presentation, using the Furstenberg theory on the product of random matrices and Monte Carlo simulations we explore the effect of randomness in the transmission performance of periodic structures designed to realize density-near-zero (DNZ) metamaterials. We show that DNZ propagation is very sensitive to randomness, and we unravel the physics behind this sensitivity with analytical theory and full-wave simulations.

2:10

2pSA4. An acoustic power divider with uniform phase and compressibility-near-zero effective material properties. Matthew S. Byrne (Elec. and Comput. Eng., Univ. of Texas at Austin, 1616 Guadalupe St., Austin, TX 78751, mbyrne@utexas.edu), Hussein Esfahlani (Photonics Initiative, Adv. Sci. Res. Ctr., New York, NY), and Andrea Alu (Photonics Initiative, Adv. Sci. Res. Ctr., Austin, Texas)

Materials with properties which can be described by a near-zero index have received much attention in the fields of microwave electromagnetics, optics, and acoustics, due to their extraordinary capabilities in wave manipulation. It was recently demonstrated, theoretically and experimentally, that acoustic media can support near-zero-index propagation, in which the effective compressibility of a waveguide channel approaches zero. In principle, this allows the complete tunneling of acoustic waves with nearly infinite wavelength (or equivalently, uniform phase). In this work, we show that these concepts can be extended to realize a novel acoustic power divider, which permits the tunneling of acoustic power to an arbitrary number of output ports, where the phase shift with respect to the input signal can be selected to be either 0 or 180 degrees. Analytical and numerical models which describe the behavior of the power divider, are presented. We conclude with an analysis that describes the limitations and trade-offs that occur due to losses as the device size is scaled.

2:25

2pSA5. Surface acoustic waves over bianisotropic metasurfaces. Li Quan (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 1616 Guadalupe St., UTA 7.215, Austin, TX 78712, liquan@utexas.edu) and Andrea Alù (Photonics Initiative, Adv. Sci. Res. Ctr., City Univ. of New York, New York, NY)

Metasurfaces are the two-dimensional planarized analogues of metamaterials, yielding unique sound manipulation along a surface. Our group recently proposed the concept of bianisotropic metasurfaces achieved by exploiting the strong nonlocal coupling between neighboring vibrating units, yielding anisotropic impedance in airborne acoustics, and we have applied this property to realize acoustic hyperbolic metasurfaces, which require extreme impedance anisotropy. In this talk, we investigate the wave propagation properties of bianisotropic metasurfaces and report all-angle backward-wave propagation on the surface. By carefully designing the coupling between unit cells, both negative phase index and energy index for surface acoustic waves are observed above the metasurface for all propagation directions, hence realizing converging waves for a radiating point source. Since in bianisotropic metasurfaces the energy can be transmitted not only through surface waves, but also through the coupling units, negative energy index for surface waves does not violate energy conservation laws, as it can be sustained by a positive energy flow within the connected unit cells composing the metasurface.

2:40–2:55 Break

2:55

2pSA6. Acoustic pressure enhancing metamaterials through impedance contrast. Hyung-Suk Kwon and Bogdan Ioan Popa (Dept. of Mech. Eng., Univ. of Michigan, University of Michigan, 2350 Hayward St, Ann Arbor, MI 48109, kwonhs@umich.edu)

Acoustic pressure enhancing devices are widely used to extend the range of sensors used for detecting weak and distant sounds. However, conventional sound enhancing devices such as acoustic horns and parabolic reflectors require bulky structures and thus limit miniaturization of sensing systems. To overcome this limitation, metamaterial techniques have been employed and promising results have been reported. However, acoustic pressure enhancing metamaterials reported so far rely on frequency dependent mechanisms to increase acoustic pressure such as resonances or wave compression methods. Therefore, in these approaches, waves are distorted during the enhancement process and this limits their applications considerably. In this presentation, we will show that metamaterials can significantly enhance acoustic waves without wave distortion while keeping the size of the metamaterial subwavelength. Our metamaterial is based on the property of the acoustic waves to increase their acoustic pressure while propagating without insertion loss from a medium of low impedance into a medium of higher impedance. The pressure gain is constant regardless of the frequency, allowing the wave to maintain its shape during enhancement. Here, we will provide the physics of the phenomenon along with numerical and experimental results which were in good agreement with the theoretical prediction.

3:10

2pSA7. Fabrication and measurement of a hybrid resonant gradient index metasurface at 40kHz. Nikhil J. Gerard (Mech. and Aerosp. Eng., North Carolina State Univ., 3141, 911 Oval Dr., EB3, Raleigh, NC 27606, rjoseph3@ncsu.edu), Huachen Cui (Mech. Eng., Virginia Tech, Blacksburg, VA), Chen Shen, Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC), Xiaoyu Zheng (Mech. Eng., Virginia Tech, Blacksburg, VA), and Yun Jing (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

Over the past decade, gradient index metasurfaces(GIMs) have been voraciously studied for the numerous wave control capabilities that they facilitate. In this regard, a hybrid structure consisting of shunted Helmholtz resonators and a straight channel is often chosen as building blocks of the metasurface. Prior research, however, has primarily focused on GIMs that operate in the audible frequency range, due to the difficulties in fabricating such intricate structures at the millimeter and sub-millimeter scales, for ultrasonic applications. In this paper, we design, fabricate and experimentally realize a gradient index metasurface for airborne ultrasound at 40kHz. The fabrication of such a GIM is made possible by projection micro-stereolithography, an emerging additive manufacturing technique capable of microscale, high aspect-ratio features over a wide area. Simulations were first conducted to verify the metasurface design. Experiments were subsequently performed to corroborate the simulations and theory. The challenges faced by the thermoviscous effects, their usefulness in certain applications and optimal designs for minimal dissipation are discussed.

3:25

2pSA8. Synchronization and avalanching amongst coupled phase oscillators, mechanical lasing. John P. Coleman and Richard Weaver (Phys., Univ. of Illinois, 1110 West Green St., Urbana, IL 61801, jpcolem2@illinois.edu)

We present theory and numerical simulations for two nonlinear systems consisting of a large number of eccentrically-weighted DC motors on mechanical bases. These are generalizations of the Kuramoto model for synchronizing phase oscillators, one chief difference being that the coupling is now frequency dependent. In one system the base is resonant and has a single degree of freedom. This system exhibits the expected second order phase transformation: for sufficient coupling strength the motors synchronize with a power output that grows with the distance above criticality. In the synchronized state the base oscillates at a single frequency (below its nominal resonance) with an amplitude that rises superluminescently with the number of motors. The degree of synchronization fluctuates intermittently, with statistics similar to those of universal crackling noise and avalanches. In our other system the motors are placed densely on a tensioned membrane, with sound speed such that wavelength is large compared to motor size and spacing. As such the structure is an active nonlinear metamaterial. As a function of coupling strength we observe a lasing transition from a near-quiescent state to a state in which the membrane is dominated by a single wavelength, and acoustic power emission is high.

3:40

2pSA9. Diffuse ultrasonic transmission between two half spaces coupled through a single glass bead. Richard Weaver and John Y. Yoritomo (Dept. of Phys., Univ. of Illinois, 1110 West Green St., Urbana, IL 61801-3080, yoritom2@illinois.edu)

We consider ultrasonics in glass blocks in contact through a single 3 mm glass bead held in place by contact forces of up to five Newtons. Hertzian contact theory predicts resonant transmission at a few isolated frequencies between 65 and 120 kHz. Resonances, based on calculations of radiative losses to the blocks, are predicted to be narrow, with widths of order 50Hz. Wide-band diffuse ultrasound from 50 to 800 kHz launched by an impulse in the upper block leads to a diffuse signal in the upper block that slowly diminishes due to absorption. It leads to a diffuse signal in the lower block that slowly increases in amplitude due to transmission—through the air and through the bead—before dissipating due to absorption. The spectrum in the lower block includes a broadband part demonstratable as due to transmission through the airgap. It also includes a part due to transmission through the bead and confined to a few isolated frequency bands. These transmission bands have widths of several kHz. We also investigate slow dynamics at the contact points by studying the change and recovery of the diffuse transmission after large amplitude conditionings.

3:55

2pSA10. A comparison study between topological insulators based on valley Hall and quantum spin Hall effects. Yuanchen Deng and Yun Jing (Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27606, ydeng5@ncsu.edu)

Over the past few years, the rapid development in the fields of condensed matter physics, electronic and photonic systems have inspired the design and experimental demonstration of various acoustic topological insulators. Among these, the topologically protected one-way propagation is a phenomenon that is gaining increased attention. Pseudospin states, which is the analogue of the Quantum Spin Hall Effect from electronic systems, has been proven to enable topological edge states in acoustics. Similarly, Valley Hall (VH) effect is also observed in acoustic systems and provides a pair of valley vortex states with opposite chirality. These valley vortex states can similarly form topologically protected edge states and in turn realize robust one-way propagation. However, the differences in the physics behind these acoustic systems give rise to distinct features such as angle selection and immunization level to various types of defects. In this article, the comparison between topological insulators (TI) and valley hall topological insulators (VHTI) address the difference and similarities in several aspects. Both of them have topologically protected bandgaps and thus the robust one-way propagation. For the maximum transmission incident angle and defect immunization, however, VH topological waveguide and TI waveguide show different characteristics.

4:10

2pSA11. Transmission enhancement through arbitrary layers with bianisotropic metasurfaces. Junfei Li, Chen Shen, and Steven Cummer (ECE, Duke Univ., 101 Sci. Dr., Rm. 3417, FCIEMAS Bldg., Durham, NC 27708, junfei.li@duke.edu)

Acoustic metasurfaces are thin, engineered structures that can control the local reflection and transmission coefficients of acoustic waves. Here, we describe our recent framework on general impedance matching with bianisotropic metasurfaces, in which elements are designed to control the local asymmetric impedance matrix. Compared with conventional quarter wavelength impedance matching layers, our proposed metasurface, with arbitrary thickness, can maximize wave transmission through an arbitrary layer with controlled phase. As a showcase, we demonstrate the design of a metasurface that helps ultrasound in water to penetrate a steel plate with almost 100% power efficiency and controlled transmission phase, which is beyond what is possible with conventional impedance matching layers. Our approach also applies to electromagnetic scenarios to help waves transmitting through walls or even metal plates.

Invited Paper

4:25

2pSA12. Flat acoustics with soft gradient-index metasurfaces. Olivier Mondain-Monval, Raj Kumar (CNRS-CRPP, Univ. of Bordeaux, 115 Av Schweitzer, Pessac 33600, France, mondain@crpp-bordeaux.cnrs.fr), Thomas Brunet (Inst. of Mech. Eng., Talence, France), Yabin Jin (Inst. of Mech. Eng., Bordeaux, France), and Olivier Poncelet (Inst. of Mech. Eng., TALENCE, France)

There is an important demand for the manufacture of materials allowing a precise control of the propagation of acoustic waves. This depends, among other things, on the speed of propagation of the waves in the material, which is expressed by the so-called acoustic index: $n = C_{\mathrm{L,ref}} / C_{\mathrm{L,mat}}$ where $C_{\mathrm{L,ref}}$ and $C_{\mathrm{L,mat}}$ are the propagation velocities of acoustic waves in a reference medium (water for us) and in the material, respectively. We have recently shown that silicone-based porous polymer materials possess exceptionally high and scalable propagation indices, which can be tuned by a precise control of the polymer's mechanical properties and porosity [1]. Our materials are obtained using an emulsion templating method coupled with a drying process that will be described in my presentation. Using this experimental approach, we were able to produce materials with an acoustic index gradient allowing, among other things, the development of a flat acoustic lens [2]. I will present the synthesis and the characterization of these materials as well as their acoustic properties. [1] A. Kovalenko, M. Fauquignon, T. Brunet, and O. Mondain-Monval, "Tuning the sound speed in macroporous polymers with hard or soft matrix," *Soft Matter* **13**, 4526–4532 (2017). [2] Y. Jin, R. Kumar, O. Poncelet, O. Mondain-Monval, and T. Brunet, "Flat acoustics with soft gradient-index metasurfaces", *Nat. Commun.* (to be published).

Session 2pSC

Speech Communication and Psychological and Physiological Acoustics: Perception and Production of Speech Directed Toward Infants and Children (Poster Session)

Mark VanDam, Cochair
Speech & Hearing Sciences, Washington State University, P.O. BOX 1495, Spokane, WA 99202

Laura Dilley, Cochair
Department of Communicative Sciences, Michigan State University, East Lansing, MI 48824

All posters will be on display from 1:30 p.m. to 4:30 p.m. To give contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 4:30 p.m.

Contributed Papers

2pSC1. Quantifying child directed speech cross-culturally across development. Melanie Soderstrom (Psych., Univ. of Manitoba, 190 Dysart Rd., Winnipeg, MB R3T 2N2, Canada, m_soderstrom@umanitoba.ca), Marisa Casillas (Max Planck Inst. for PsychoLinguist, Nijmegen, The Netherlands), Elika Bergelson (Duke Univ., Durham, NC), Jessica Kirby (Psych., Univ. of Manitoba, Winnipeg, MB, Canada), Celia Rosemberg, Alejandra Stein (CIIPME Conicet, Buenos Aires, Argentina), Anne Warlaumont (UCLA, Los Angeles, CA), and John Bunce (Psych., Univ. of Manitoba, Winnipeg, MB, Canada)

Child-directed speech (CDS) influences language development (e.g., Golinkoff *et al.*, 2015), but varies across cultural and demographic groups (Hoff, 2006). Recent work examining speech heard by North American English (NAE) infants found an increased *proportion* of CDS with age (Bergelson *et al.*, 2018). Quantity of CDS remained relatively constant across age, while quantity of adult-directed speech (ADS) decreased. We replicate these findings using a different methodology, and expand them to include other language communities. Our data come from daylong audio recordings of 58 children ages 2–36 months from the ACLEW dataset (Bergelson *et al.*, 2017; 30 children acquiring NAE, 10 UK English, 8 Argentinian Spanish, and 10 Tseltal/Mayan). Ten randomly selected 2-min segments (Tseltal: nine 5-min segments) from each child were annotated for speaker gender, age (child or adult), and addressee for each utterance. We calculated the minutes per hour of CDS, ADS, and all speech. Preliminary analyses find high variability in overall language input across individuals, age, and culture, and partially replicate the Bergelson *et al.* (2018) pattern of results. Ongoing annotation will permit finer-grained analyses of sub-group differences. Further analyses will examine the influence of factors such as speaker gender, number of speakers, and maternal education.

2pSC2. Longitudinal development of conversational exchanges in children with hearing loss. Mark VanDam and Rebecca Hibben (Speech & Hearing Sci., Washington State Univ., P.O. Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu)

Conversations between adults and children have been shown to be an important factor affecting the development of cognition, academic performance, and language use in children. Conversational exchange frequency between parents and children has been shown to increase with age. It has also been demonstrated that degraded sensory input, such as with hearing loss, can result in reduced language, speech, and communication skills in children. This study examines the longitudinal development of conversational exchanges between children with hearing loss and their parents. Thirty-nine families contributed 370 daylong recordings constituting 4337 h of audio collected from a body-worn audio recorder. Automatic speech processing techniques were used to identify and tally conversational exchanges. The main longitudinal effect of age on conversation exchange rate is confirmed, but the effect is weaker and may have a different developmental trajectory for families of children with a hearing loss. This result may be influenced by differences in the development of joint attention between children who are typically developing and those with hearing loss.

2pSC3. Speech perception in children with reading disabilities: Phonetic perception is the problem. Yashuo Wu (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 512 E Green St. Apt. 301, Champaign, IL 61820, yashuow2@illinois.edu), Jie Lu (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Zhanjiang, China), Joseph C. Toscano (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL), Cynthia Johnson (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Champaign, IL), and Jont b. Allen (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Reading disability (RD) is widely accepted as a key obstacle in the development of literacy. Studies show that 15–20% of grade-school students are RD. Many quit high-school and go to jail. We shall show that RD for 8–12 yrs is related to inadequate phonetic identification ability, rooted in preschool language development. We used two tests (10 thousand responses/child): (1) A 3-interval forced choice procedure (Syllable Confusion Oddball Task: SCO). (2) A single CV/VC presentation task with oral response, to label CV/VC phones (Nonsense Syllable Confusion Matrix: NSCM). The experimental results showed that for the SCO task the 10 RD cohort had, on average 5 times the error compared to the 6 RC reading control (RC) cohort. The errors were highly idiosyncratic, analyzed by logit. (1) RDs have significant speech perception problems, despite normal pure-tone hearing and language ability. (2) When comparing the SCO and NSCM results, our findings are consistent with a reduced ability to label CV/VC sounds presented in random temporal order. This seems consistent with phone memory dysfunction. (3) These conclusions are at odds with previous studies finding no indication of phone identification impairment.

2pSC4. Preschoolers initiate more conversations than their parents. Mark VanDam, Sarah Campanella, Kiley Wolfenstein (Speech & Hearing Sci., Washington State Univ., P.O. Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu), Paul De Palma (Comput. Sci, Gonzaga Univ, Spokane, WA), and Daniel Olds (Speech & Hearing Sci., Washington State Univ., Spokane, WA)

Children develop, learn, and refine the complex rules of how to have a conversation beginning in their preschool years. Recent work on conversational exchanges and language usage within families has shown differences (and in some cases, similarities) in how mothers, fathers, girls, and boys interact and converse with each other. It has been suggested that mothers contribute more child-oriented and child-driven exchanges, fathers contribute more formal language that includes problem-solving and linguistic manipulation, girls seek to maintain relationships, and boys seek to establish dominance and attract or maintain and audience. These factors may influence the roles each interlocutor plays in a communicative exchange. One aspect of verbal interaction is who initiates a conversational exchange. This study examined 134 daylong audio recordings using automated speech processing techniques to estimate the frequency of conversation initiation for mothers, fathers, girls, and boys. We found that children initiate conversations most frequently, followed by mothers, followed by fathers. We found no rate difference between girls and boys. Results are consistent with the Bridge Hypothesis or Apprenticeship Model in which interlocutors are motivated in part by their social role, in this case by the social modeling parents demonstrate for their children.

2pSC5. The development of emotional speech prosody perception in 3- to 14-month infants: a preferential listening study. Chieh Kao and Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, kaoxx096@umn.edu)

Developmental studies have shown strong evidence that socially enriched speech signal (including prosodic modifications) attracts infants' attention and facilitates language development. While emotion understanding is evident at 9 months of age (Otte *et al*., 2015), the developmental trajectory of the emotional speech prosody perception is still unclear. The present study adopted a widely used preferential looking paradigm to measure 3- to 14-month-old infants' listening preference to English words spoken in neutral, happy, angry, and sad tones. Analysis using a linear mixed model showed that infants' preference of emotional prosody changed as a function of age. On average, the three-month olds listened longer to all emotional prosodies over the neutral one whereas older infants showed significantly diminished interests in the sad prosody, followed by the happy and angry voices. Around 12 months, infants appeared to listen to emotional prosodies equally with the exception of reduced interest in the angry prosody. These preferential listening measures were not correlated with the varying durations or fundamental frequencies of the spoken words for the different emotional categories, indicating that the development of emotional speech prosody is not purely driven by the acoustical properties but rather involves higher-order social cognition.

2pSC6. Syllable production in hard-of-hearing preschoolers. Mark VanDam, Kiley Wolfenstein, Sarah Campanella (Speech & Hearing Sci., Washington State Univ., P.O. Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu), Daniel Olds (Comput. Sci, Washington State Univ, Spokane, WA), and Paul De Palma (Comput. Sci, Gonzaga Univ., Spokane, WA)

Expressive language in preschoolers has been shown to be positively related to later language development, although not all studies have shown robust effects. Most studies that consider sex have shown that girls develop expressive language earlier than boys. It has also been widely demonstrated that children with a variety of speech-, language-, and hearing-related disorders show deficits in expressive language skills. Children with hearing loss in particular have shown deficits in consonant production, prosodic control, vowel duration, and mean length of utterance. This study examines the expressive language of preschool children with hearing loss using estimates of syllable production from 366 daylong audio recordings totaling over 4348 h of audio processed with unsupervised automatic speech processing techniques. We found no main effect of sex, but typically-developing children were more voluble on average. Unexpectedly, an interaction effect suggests that typically developing boys may be driving the observed differences. This work adds to the evidence of social and biological variability in the speech production of children, and is further proof-of-concept of developmental speech production work using massive data sets and automatic analysis methods.

2pSC7. Child-directed speech enhances preschoolers' speech perception in noise. Nicholas A. Smith (Dept. of Speech, Lang. and Hearing Sci., Univ. of Missouri, Columbia, MO 65211, smithnich@health.missouri.edu), Christine A. Hammans, Timothy J. Vallier (Boys Town National Res. Hospital, Omaha, NE), and Bob McMurray (Dept. of Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

A dominant idea in research on infant-directed speech is that caregivers hyperarticulate their speech and exaggerate speech clarity as a means of facilitating language learning for infants. Much less is known about the acoustic-phonetic properties of speech to young children who are more amenable to perceptual testing. This study tests whether child-directed speech (CDS) is more intelligible by testing 4- to 6-year-old listeners in a speech-in-noise task in which they selected the corresponding picture on a touchscreen. Target words were two-way minimal pairs that contrasted in terms consonant voicing (e.g., "back" versus "pack"), vowel (e.g., "back" versus "beak"), or consonant and vowel (e.g., "back" versus "peak"). At a constant signal-to-noise ratio of -6 dB, speech recognition performance was significantly greater for CDS than for adult-directed speech (ADS). However, significant differences were also found across stimulus talkers, with some mothers providing a greater overall CDS benefit than others. Furthermore, CDS was related to greater enhancement of vowel contrasts in some mothers, and greater enhancement of consonant voicing in others. These perceptual results are discussed in terms of the acoustic-phonetic properties of these speech productions.

2pSC8. Contextual influences on infants' attention to child-directed speech. Robin Panneton, Tyler McFayden, Madeleine Bruce, and Caroline Taylor (Dept. of Psych., Virginia Tech, Blacksburg, VA 24061-0436, panneton@vt.edu)

Initial publications of infants' preference for infant (child)-directed speech (CDS) over adult-directed speech (ADS) produced a swift generalization to all early development. In fact, a recent meta-analysis found that the CDS preference to be robust (Dunst *et al*., 2012) and the ManyBabies I replication effort extends this view across labs, methods, and samples (Frank *et al*., 2018). Nonetheless, there are important demonstrations of how this preference gets attenuated and augmented by various factors. For example, Cooper *et al*. (1997) found that 1-mo-olds did not prefer CDS over ADS when both recordings were of the infants' own mothers, but did prefer CDS when the speakers were unfamiliar women. Other studies on CDS perception find influences of the language being spoken, the modality of presentation, the age of the infant, the clinical status of the mother and/or of the infant. The influence of context on infants' attention to speech is an under-appreciated and understudied aspect of early language learning. This presentation will summarize contextual moderators of the CDS preference that emanate from the infant and from the infant's developmental milieu, and offer suggestions for other contexts that most likely impact infants' attention to CDS but have yet to be investigated (e.g., poverty).

2pSC9. Using acoustic features of mothers' infant-directed speech to predict changes in infant biobehavioral state. Jacek Kolacz (Traumatic Stress Res. Consortium at the Kinsey Inst., Indiana Univ., Lindley Hall 428, 150 S Woodlawn Ave., Bloomington, IN 47405, jkolacz@iu.edu), Elizabeth B. daSilva (Indiana University-Purdue Univ. Columbus, Columbus, IN), Gregory F. Lewis (Intelligent Systems Eng., Indiana Univ., Bloomington, IN), Bennett I. Bertenthal (Dept. of Psychol. and Brain Sci., Indiana Univ., Bloomington, IN), and Stephen W. Porges (Traumatic Stress Res. Consortium at the Kinsey Inst., Indiana Univ., Bloomington, IN)

Infant directed speech is marked by exaggerated frequency modulation and strong high frequency power, features that may provide physiological cues for mobilization or calming (Porges and Lewis, 2010; Kolacz *et al*., 2018). We examined whether these features predicted changes in infant

biobehavioral state during the Still Face Paradigm, a stressor in which the mother withdraws and reinstates social cues. 98 mother-infant dyads participated when infants were 4-8 months old. Infant heart rate and respiratory sinus arrhythmia (a measure of cardiac parasympathetic control) were derived from an electrocardiogram (ECG). Infant behavioral distress was measured by vocal, facial, and body movement distress. Mothers' vocalizations were measured using spectral analysis within designated frequency bands and modulation using a 2-dimensional fast Fourier transform of the audio spectrogram. Maternal frequency modulation predicted decreases in infant heart rate ($p = .030$), mid-frequency acoustic power (500–5000 Hz) predicted increases in cardiac parasympathetic regulation in infants with low parasympathetic tone ($p = .024$), and high frequency power predicted increases in infant behavioral distress in infants who were not initially distressed ($p = .011$). These results suggest that mothers' vocal frequency band power and modulation may be aspects of infant-directed speech that are relevant for regulating infant biobehavioral state.

2pSC10. The effects of parent coaching on language outcomes at 18 and 24 months: A randomized controlled trial. Naja Ferjan Ramirez, Sarah Lytle, Ruofan Cai, and Patricia K. Kuhl (Univ. of Washington, Box 357988, Seattle, WA 98115, naja@u.washington.edu)

The prevalence of parentese in speech directed to 11- and 14-month-old infants predicts infants' concurrent babbling as well as their future language skills at 24 months, suggesting that this speaking style may enhance learning. We recently showed that when parents are "coached" about the importance of language input to infants, and parentese, they increase the proportion of parentese and child-directed speech. This has an immediate and positive effect on child language outcomes at 14 months. In the present study, we asked whether the effects of coaching parents extend to longer-term language outcomes. Families of typically developing 6-month-old infants were assigned to Intervention (parent coaching) and Control (no coaching) groups. Parent coaching took place when infants were 6- 10-, 14- and 18-months of age, and included quantitative and qualitative linguistic feedback derived from each family's first-person LENA recordings at home. Language outcomes were measured at 18- and 24-months of age. Parent coaching significantly enhanced the percentage of child-directed speech and parentese in parental input between 6 and 18 months in coached vs. uncoached parents. Children of parents who received coaching showed enhanced language outcomes at 18 and 24 months.

2pSC11. The real-time dynamics of child-directed speech: Using pupillometry to evaluate children's processing of natural pitch contours. Mira L. Nencheva, Elise A. Piazza, and Casey Lew-Williams (Dept. of Psych., Princeton Univ., Princeton, NJ 08540, nencheva@princeton.edu)

Young children prefer child-directed speech (CDS) to adult-directed speech (ADS) (Cooper and Aslin 1990), and its structural and prosodic features are known to facilitate learning (Thiessen *et al*., 2005; Graf Estes and Hurley, 2013). However, little is known about how the real-time dynamics of CDS prosody affect young children's engagement and learning. In Experiment 1, we evaluated moment-to-moment processing of pitch variation using measures of pupil size synchrony across children (Kang and Wheatley, 2017). 24-to-30-month-old children listened to a story in CDS and ADS, and we found that pupil synchrony was higher for CDS than ADS. Next, using hierarchical clustering, we uncovered 4 main word-level pitch contours from a natural CDS corpus and identified contours (specifically, U- and inverted-U-shaped contours), which elicited lower vs. higher synchrony, respectively. In Experiment 2, we found that children learned novel words better when they were presented in higher- vs. lower-synchrony contours. Importantly, synchrony for a novel word during training significantly predicted learning for the same word at test. By revealing a physiological response that is sensitive to the real-time dynamics of prosody, this investigation yields a new subsecond framework for understanding children's engagement with a signal known to support early language learning.

2pSC12. The effects of parental interaction on French-English bilingual infants' vocalization and turn-taking rates. Katherine Xu, Adrial J. Orena, Yufang Ruan, and Linda Polka (School of Commun. Sci. & Disord., McGill Univ., 2001 Mcgill College Ave., 8th Fl., Montreal, QC H3A 1G1, Canada, tian.y.xu@mail.mcgill.ca)

Prior studies suggest that parent-infant interactions play an important role on infant volubility, which is an important indication of language development. However, little is known about how speaker and language contexts affect infant volubility in bilingual infants. In the current study, we examined how the speaker context (Mother versus Father) and the language context (French versus English) might influence bilingual infants' rate of vocalization and turn-taking. We analyzed naturalistic daylong recordings from English-French bilingual parent/infant dyads ($n = 21$). Preliminary analysis showed that mothers elicited more turn-taking in their 10-month-old infants than fathers did, but they did not elicit more vocalizations. Importantly, there was no effect of language on infant vocalization or turn-taking. We will alsoreport analyses that examine whether the dominant language in the bilingual child's input has an effect on infant volubility that is independent of talker effects. These novel findings will inform our understanding of how parent-infant interactions shape the language development of infants being raised in bilingual families.

2pSC13. Acoustic correlates of hypokinetic articulation of continuously spoken sentences. Amitava Biswas, Mary Schaub, and Steven Cloud (Speech and Hearing Sci., Univ. of Southern Mississippi, 118 College Dr. #5092, SHS Lab., Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Hypokinetic speech involves reduced range of motion of the articulators such as tongue, lips, and jaw. As a result, the formant transitions are relatively reduced. This characteristic may be observed in some adults and children. An algorithm has been developed to estimate the relative variations in formant transitions in continuously spoken sentences. Its sensitivity, specificity, and applications in clinics will be discussed.

2pSC14. Infant-directed speech register in children with and without hearing loss. Maria V. Kondaurova, Kaelin Kinney, Abigail Betts (Psychol. & Brain Sci., Univ. of Louisville, 317 Life Sci. Bld., Louisville, KY 46292, maria.kondaurova@louisville.edu), Lindsay Nolan (Heuser Hearing Inst. & Lang. Acad., Louisville, KY), and Mark VanDam (Elson S. Floyd College of Medicine, Dept. Speech & Hearing Sci., Washington State Univ., Spokane, WA)

Do children with hearing loss use infant-directed speech? The study examined speech characteristics of a 6-year-old child with bilateral cochlear implants and an age-matched child with normal-hearing while interacting with their infant siblings (age 29 and 20 months) and with their mothers. Child-sibling and child-mother interactions were recorded in two conditions. In the "toy" condition, the children explained to their siblings and their mothers how to assemble a toy. In the "book" condition, the children narrated a story using a picture book. Sixty-five vocalizations from each child's speech sample were extracted in each condition. Mean fundamental frequency, fundamental frequency range, utterance duration, number of syllables per utterance, and speech rate were measured. Both children produced higher fundamental frequency, expanded fundamental frequency range, shorter utterance duration, and slower speech rate in the sibling- compared to mother-directed speech in both the "book" and "toy" conditions. For the mother-directed speech only, the children produced lower fundamental frequency, longer utterance duration and more syllables per utterance in the "book" than the "toy" condition. The results suggest that children with and without hearing loss modify prosodic characteristics of their speech when interacting with a younger sibling but the strength of the modification may be task-dependent.

2pSC15. Acoustical regularities in infant-directed vocalizations worldwide. Cody J. Moser (Dept. of Anthropology, Texas A&M Univ., 340 Spence St., College Station, Texas 77840, mose3774@tamu.edu), Harry Lee-Rubin (Dept. of Psych., Harvard Univ., Cambridge, MA), Infant-Directed Vocalizations Collaboration (none, Cambridge, MA), Constance M. Bainbridge, Stephanie Atwood, Max M. Krasnow, and Samuel A. Mehr (Dept. of Psych., Harvard Univ., Cambridge, MA)

Adults often differentiate their song and speech between infants and other adults. Why? Is this a product of just some cultures or does it reflect a universal part of human vocal communication and human cognition? On the latter hypothesis, infant directed-song might regularly exhibit certain features across cultures, including high redundancy and repetition, high signal-to-noise ratios, and superb vowel prolongation and stability compared to adult-directed song. We built a corpus of 1614 recordings of infant- and adult-directed singing and speech produced by 411 people living in 20 societies, including hunter-gatherers and subsistence farmers. Each participant provided examples of each of the four vocalization types. Using exploratory and confirmatory analyses, we show that the acoustical features of infant-directed song and speech are universally distinct from adult-directed song and speech, especially in terms of the phonetic space of their formants, their general rhythmic structure, and their pitch range attributes.

2pSC16. Separability of infant-directed from adult-directed speech is affected by number of channels in cochlear-implant simulated speech. Meisam K. Arjmandi, Laura Dilley (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd. Oyer Speech & Hearing, Rm. 211A, East Lansing, MI 48824-1220, khalilar@msu.edu), Yuanyuan Wang (Ohio State Univ., Columbus, OH), Mario Svirsky (New York Univ., New York, NY), Matt Lehet (Communicative Sci. and Disord., Michigan State Univ., Pittsburgh, Pennsylvania), and Derek Houston (Ohio State Univ., Columbus, OH)

Many studies have demonstrated benefits of infant-directed speech (IDS) over adult-directed speech (ADS) for language development. For deaf children who use cochlear implants (CIs), a variety of factors might reduce the advantage of IDS in language development. We hypothesized that spectral degradation due to the number of channels in CI processing negatively affects acoustic separability of IDS and ADS, potentially reducing benefits for language development of IDS in early childhood. 493 sentences spoken in two speaking styles (IDS and ADS) were processed using 4, 8, 12, 16, 22, and 32 channels noise-vocoders to simulate the spectral degradation caused by CIs. The sentences were partitioned into frames of 30 ms and represented by Mel-frequency cepstral coefficients. The Mahalanobis distance was used to calculate the acoustic distance across speaking styles (ID versus AD) and processing condition (4, 8, 12, 16, 22, and 32 channels). The results show that spectral degradation imposed by CI processing has significant negative effects on separation of IDS from ADS. These findings suggest that the spectral information in speech received by infants with CIs is substantially degraded, which can weaken the facilitative link between IDS and language development in this at-risk population. [Work supported by NIH grant R01DC008581.]

2pSC17. Infant-directed speech enhances recognizability of individual mothers' voices. Thayabaran Kathiresan (Univ. of Zurich, Zurich, Switzerland), Laura Dilley (Michigan State Univ., East Lansing, MI), Simon Townsend (Univ. of Zurich, Zurich, Switzerland), Rushen Shi (Universite du PQ a Montreal, Montreal, QC, Canada), Moritz Daum (Univ. of Zurich, Zurich, Switzerland), Meisam K. Arjmandi (Michigan State Univ., East Lansing, MI), and Volker Dellwo (Phonet. Lab., Univ. of Zurich, Plattenstrasse 54, Zurich 8005, Switzerland, volker.dellwo@uzh.ch)

Adult speakers commonly alter their voices when talking to infants, giving rise to an infant-directed speech (IDS) style. Here we tested the effects of infant-directed speech on the recognizability of a speaker's voice. 10 Swiss-German mothers were recorded talking to their infants IDS and talking to an adult experimenter (in adult-directed speech, ADS). We studied the indexical properties using Mel-frequency cepstral coefficients (MFCCs). By using an unsupervised K-means clustering algorithm, the segmental 13-dimensional MFCCs were clustered and reduced to two dimensions using Principal Component Analysis. Results showed that the relative area of IDS occupied in the 13-dimensional space was significantly larger compared to the area occupied by ADS. A supervised language-independent Gaussian Mixture Model revealed that this expansion benefitted the recognizability of mothers' voices. This means that the higher indexical variability in IDS fosters recognition of individual mothers. Results are consistent with the view that IDS may have evolved in part as a strategy to promote indexical signaling by a mother to her offspring, thereby promoting mother-infant attachment and fostering offspring survival.

2pSC18. Vowel space and variability in infant- and adult-directed speech. Kelly D. Burkinshaw (School of Lang., Linguist, Literatures and Cultures, Univ. of Calgary, Craigie Hall C211, 2500 University Dr. N.W., Calgary, AB T2N 1N4, Canada, kburkins@ucalgary.ca), Lori L. Holt (Dept. of Psych., Carnegie Mellon Univ., Pittsburgh, PA), and Suzanne Curtin (Dept. of Psych., Univ. of Calgary, Calgary, AB, Canada)

Infant- and adult-directed speech are acoustically-distinct registers, but whether the characteristics of infant-directed speech (IDS) promote speech category learning is unclear. Several studies have reported an expanded vowel space in IDS compared to ADS; point vowels (/i/, /a/, /u/) are, on average, more distinct from each other in IDS. But, other studies report greater intra-category variability which might diminish any benefits of an expanded vowel space. Here, we examined vowel productions across five vowels as mothers spoke to their infants (7- or 15-months) in IDS and an adult experimenter in ADS. We observed an expanded point vowel space and an increase in variability of within-category vowel productions in IDS toward 15-month-olds, compared to ADS. Thus, although the centroids of the point vowels were more separated in IDS toward 15-month-olds than ADS, production variability led to substantial category overlap. Yet, classification modeled using Discriminant Function Analysis (DFA) revealed near-ceiling classification rates for both registers, indicating that there was no classification advantage of the increased distance among IDS point vowels. We found neither vowel space expansion nor increased variability in IDS toward 7-month-olds. These findings inform how distributional characteristics of speech input may contribute to vowel category learning in infancy.

2pSC19. Investigating naturalistic code-switching directed towards infants. Lena V. Kremin (Psych., Concordia Univ., 7141 Sherbrooke St. W, PY-033, Montreal, QC H4B 1R6, Canada, lena.kremin@mail.concordia.ca), Adriel John Orena (School of Commun. Sci. and Disord., McGill Univ., Montréal, QC, Canada), Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada), and Krista Byers-Heinlein (Psych., Concordia Univ., Montreal, QC, Canada)

Mixing two languages in speech (i.e., code-switching) is prevalent in multilingual settings, including in speech directed towards infants. Prior research suggests a link between parental code-switching and vocabulary size (Byers-Heinlein, 2013). Moreover, laboratory work suggests that some types of code-switching appear more difficult for infants to process than others (Byers-Heinlein *et al*., 2017; Potter *et al*., 2018). This raises the possibility that the effects of parental code-switching depend on the parents' specific behavior in terms of the frequency, location, and purpose of code-switching (Byers-Heinlein, 2017). Prior studies of parental code-switching relied on self-report or short lab observations. In this study, we analyze parental code-switching behavior in a corpus of daylong home recordings of 21 infants (at 10- and 18-months) from French-English bilingual families in Montréal. We will identify instances of parental code-switching, their syntactic location, the direction of the switch, and the apparent reason for the switch (e.g., teaching vocabulary, translating an entire utterance). Preliminary results indicate that the frequency of code-switching varies between families and that code-switching between sentences is more common than code-switching within a sentence. This project will provide the first in-depth investigation about the characteristics of naturally produced parental code-switching.

2pSC20. Average daily speech exposure for fetuses. Brian B. Monson and Molly Cull (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61822, monson@illinois.edu)

The quality and quantity of speech and language exposure during early childhood is believed to be predictive of language ability during later childhood development. However, the human auditory nervous system comes "online" *in utero,* at least as early as 23 weeks' gestation. It has been demonstrated that intrauterine fetal experience with extrauterine sounds during this last trimester of gestation is sufficient to impact auditory brain development and neural responses to speech. Whether fetal exposure to speech affects later childhood language development remains an open and difficult question. To begin to address this question, we collected and analyzed fetal auditory exposure data for sounds generated in the extrauterine environment using small audio recorders worn by pregnant women during the third trimester of pregnancy. Averaged across 2200 h of audio data, daily speech exposure for individual fetus subjects ranged from 2.8 to 5 h, suggesting some newborns may begin extrauterine life with less than 60% of the speech and language exposure of their peers. Whether this variability is associated with variability in subsequent development remains to be seen.

2pSC21. Context-dependent hyperarticulation of the Korean three-way laryngeal stop contrast in clear speech. Seulgi Shin and Allard Jongman (Linguist, Univ. of Kansas, 1846 Tennessee St., Apt. #4, Lawrence, KS 66044, seulgi.shin@ku.edu)

We investigate whether and how speakers' adaptation to feedback is reflected in hyperarticulation by comparing the productions of Korean aspirated, lenis, and fortis stops in plain versus clear speech. Ten Seoul Korean speakers were asked to read a stop-initial nonword and repeat the nonword after receiving misrecognition feedback that contained a contrasting stop. f0 and VOT were examined as main cues to the Korean stop distinction. Results showed that VOT was lengthened for aspirated and lenis stops in clear compared to plain speech. This difference in VOT was attributable to speakers' modification in response to different types of feedback. VOT for aspirated stops was lengthened after both fortis and lenis feedback whereas VOT for lenis stops was lengthened only after fortis feedback. f0 did not show a difference between plain and clear speech. However, f0 enhancement for lenis and fortis stops depended on the type of feedback. f0 for lenis stops was lowered after aspirated feedback and raised after fortis feedback. Fortis stops' f0 was lowered after aspirated and raised after lenis feedback. Overall, speakers can make local modifications in VOT and f0 in response to specific feedback, suggesting precise control in their effort to distinguish sounds and maintain intelligibility.

2pSC22. Characterising maternal pitch contours used during interactions with infants at high and low risk for autism spectrum disorder. Alix Woolard (Psych., Univ. of Newcastle, University Dr., Callaghan, NSW 2308, Australia, alix.woolard@uon.edu.au), Alison Lane (Health Sci., Univ. of Newcastle, Callaghan, NSW, Australia), Linda Campbell, Frini Karayanidis (Psych., Univ. of Newcastle, Callaghan, NSW, Australia), Daniel Barker (Medicine and Public Health, Univ. of Newcastle, New Lambton, NSW, Australia), Larissa Korostenski (Neonataology, John Hunter Children's Hospital, New Lambton, NSW, Australia), Shelly Lane (Health Sci., Univ. of Newcastle, Callaghan, NSW, Australia), and Titia Benders (Linguist, Macquarie Univ., North Ryde, NSW, Australia)

Infant-directed speech (IDS) is the speech register used when interacting with infants. Pitch contours are a salient aspect of IDS and facilitate infant language and socio-communicative development. Little research investigates pitch contours within the context of socio-communication or language deficits, such as infants at high-risk (HR) for Autism Spectrum Disorder (autism). The aim of this study was to characterise pitch contours used by mothers when interacting with HR infants compared to mothers interacting with low-risk (LR) infants. 18 mothers and their 12-month-old infant (12m, 6f) participated in 15-minute recorded interactions. Autism risk was assessed via parent and observer-report. Pearson product-moment correlations were performed to determine relationships between maternal pitch contours and autism risk. Increased risk for autism was associated with fewer utterances (r = -0.576, N = 18, p = 0.01) and fewer rising (r = -0.586, N = 18, p = 0.01), sinusoidal (r = -0.636, N = 18, p = 0.005), and flat contours (r = -0.679, N = 18, p = 0.01), and more complex (r = 0.584, N = 18, p = 0.01), rapid (r = 0.526, N = 18, p = 0.03), and u-shaped contours (r = 0.619, N = 18, p = 0.02). These preliminary data suggest that mothers of HR infants use different patterns of pitch contours than LR mothers. Further assessment of IDS used with HR infants is warranted to identify at what stage IDS patterns deviate between groups.

Session 2pSP

Signal Processing in Acoustics, Acoustical Oceanography, Physical Acoustics, Underwater Acoustics, and Structural Acoustics and Vibration: Borehole Acoustics Logging for Hydrocarbons Reservoir Characterization

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Chair's Introduction—1:30

Invited Papers

1:35

2pSP1. Logging while drilling shear sonic logging in very large boreholes. Matthew Blyth (Schlumberger, 32702 Whitehaven Pl., Fulshear, TX 77441, mblyth@slb.com), Naoki Sakiyama (Schlumberger, Sagamihara, Japan), Hiroaki Yamamoto (Schlumberger, Houston, TX), Atsushi Oshima (Schlumberger, Sagamihara, Kanagawa, Japan), and Eduardo Saenz (Schlumberger, Houston, TX)

As LWD sonic tools have become more advanced, they have become more routine in use and are now commonly used in multiple hole sections, with the results having applications across a wide range of problems in well construction. The acquisition of quadrupole shear measurements in large boreholes and slow formations is highly challenging however, due to the borehole conditions, the very slow formation shear values and the effects of the drilling fluid. This paper will discuss the limits of the measurement under these conditions, using both modeled data and real well examples and the effect on measurement quality as a result of these conditions will be shown. The modeling studies shown indicate the importance of understanding the drilling mud properties (slowness and density) was well as the formation and borehole properties when attempting to derive robust shear information under these conditions. The results show that, although the acquisition of reliable shear data in these conditions is challenging, it is not impossible, provided that the LWD tool is correctly designed, the physics are understood and suitable processing applied.

1:55

2pSP2. Simulations of collar waves for acoustic logging while drilling in the frequency and the spatial domains. Xiao He, Yunjia Ji, Hao Chen, and Xiuming Wang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., 21, Northern 4th Ring Rd. West, Haidian District, Beijing 100190, China, hex@mail.ioa.ac.cn)

Tool waves generated by the presence of the drill collar can cover the formation signals we need. The suppression for those collar waves has been a challenge for designing acoustic logging while drilling (ALWD) tools. To study the collar wave characteristics, we evaluate the excitation intensity of the modes with varying frequencies and positions by solving the elastic wave equations. Analysis results show that the collar wave energy concentrates in the cylindrical tool model. And the radial location of its peak gradually moves from the inner surface to the outer wall with the increasing frequency. The particle vibration trajectories contributed by the collar wave are also investigated. It is revealed that the collar mode in the lower frequency range has an approximately longitudinal polarization, while it tends to the transverse-wave-like motion at high frequencies. According to these features, we propose strategies on the tool structure designs to suppress the collar waves in the lower- and the higher- frequency ranges, respectively. Numerical examples further validate that the proposed collar structures with interlayers or grooves can well attenuate the collar wave propagation.

2:15

2pSP3. Determining fracture properties from acoustic borehole waves: theory and experimental results. Huajun Fan (College of Geophys. and Information Eng., China Univ. of Petroleum (Beijing), 18 Fuxue Rd., Changping, Beijing 102249, China, hj.fan@hotmail.com) and David Smeulders (Dept. of Mech. Eng., Eindhoven Univ. of Technol., Eindhoven, The Netherlands)

A borehole in the subsurface may penetrate rocks which are porous, permeable and fractured. Pressure transients in the borehole will therefore cause viscous fluid to flow into and out of the wall of the borehole. This forced flow consumes some energy and affects the phase velocity and amplitude of the waves traveling in the borehole fluid column. These effects can be evaluated to extract information about the rocks adjacent to the borehole, which is of paramount importance for the oil and gas industry. This work discusses laboratory wave experiments in a 7.5 m long vertical shock tube. Rock samples having a vertical borehole and horizontal fractures are installed in the shock tube and filled with water. The shock tube generates broadband pressure transients in the borehole. These are measured in the borehole at variable depth by means of a sliding pressure probe. Repetitive experiments are combined into borehole microseismograms.

In this way borehole wave reflection and transmission coefficients over the fractures are determined. The results show good agreement with low-frequency theory, where it is assumed that the wavelengths are much larger than the tube diameter. Inversely, it is shown that the aperture and the length of the fracture can directly be inferred from acoustic borehole experiments.

2:35

2pSP4. Features of the protective cover of the ultrasonic imager with high resolution. Kamil Yusupov, Victor Kosarev, and Adel Akchurin (Kazan Federal Univ., 18 Kremlyovskaya St., Kazan 420008, Russian Federation, kamil.usupov@kpfu.ru)

This work describes the features and design of the acoustically semitransparent cover of a borehole ultrasonic imager with high resolution. This scanner is designed to investigate the fine structure of the well surface by ultrasonic sounding method at a frequency of 800 kHz. The ultrasonic transducer is rotating around a central axis. However, in the design of the device fixed semitransparent protective cover is provided for prevent damage of mechanical parts. Nevertheless, during reflected signals recording from the wall of well the undesirable noise reflections appear from protective cover. These noises interfere with the algorithm of detecting the main signal. The logging tool electronics is based on a single FPGA chip, its logic cells capacity allows determining the maximum reflection amplitude and its location, as well as recording the signal time window to the SD-card (with 32GB storage space). This windows are processed at the personal computer after logging. Such approach allows analyzing waveforms and noises. After a multiple laboratory and well experiments this logging tool was allow improving the design of the acoustically semitransparent protective cover for maximum attenuation of the reflection noise. In addition, the forms of the recorded acoustic signals and the design of the protective cover are given.

2:55

2pSP5. Assessing organic richness of source rocks through integration of acoustic logs and microresistivity images. M. S. McQuown (Weatherford, 10844 Diane Dr., Golden, CO 80403, scott.mcquown@weatherford.com)

Accurate evaluation of source rock thickness and organic richness are key components to estimating the hydrocarbon potential of a basin and identifying target zones in self-sourcing shale plays. The millimeter-scale variability in mineralogy and organic matter concentrations in source rocks can make petrophysical quantification problematic when relying on the resolution of standard well logs. Integrating acoustic well log data with microresistivity images with can improve resolution while generating more meaningful information than their separate results. We developed a new algorithm that utilizes the superior resolution and circumferential coverage of microresistivity images and integrates them with enhanced acoustic data to assess the organic richness of source rocks. The result is an oriented 360 deg pseudo-sonic image that reveals anomalous properties related to organic richness. The validity of the method is assessed by comparing the results with standard well log-derived solutions for total organic carbon. Excellent correlation between the image results and petrophysical solutions is observed.

Contributed Papers

3:15

2pSP6. A method to locate acoustic emission events induced by hydraulic fracturing using waveform correlation weighted by amplitude stacking. Chengwei Zhang, Wenxiao Qiao, and Xiaohua Chen (College of Geophys. and Information Eng., China Univ. of Petroleum-Beijing, No. 18, Fuxue Rd., Changping District, Beijing 102249, China, zhangchengwei.cq@163.com)

Acoustic emission monitoring has been the best technique to evaluate hydraulic fracturing which is the most common method used to induce fractures for stimulating hydrocarbon reservoirs. Correlation-based and amplitude-stacking-based methods, both are based on migration and are all suitable for locating acoustic emission events induced by hydraulic fracturing without the need to pick the arrival times of the P- and S-waves. By comparing and analyzing the advantages and disadvantages of the two location methods, we propose a new weighted correlation method using both amplitude and waveform correlation. First, we calculate the travel time of acoustic waves from the trial point in the formation to each receiver by the ray-tracing method, and further to determine the time-window positions of the P- and S-waves on all waveforms. Then, we calculate the correlation of the waveforms in the windows and the amplitude stacking of the average energy ratio between the short-time window and the long-time window on the original acoustic waveforms. Finally, we use the correlation weighted by amplitude stacking to image space locations of acoustic emissions. Tests with synthetic data show that the weighted correlation method has stronger stability and lower location uncertainty than the existing migration-based location methods.

3:30

2pSP7. Borehole sonic array processing and the group versus phase velocity debate. Said Assous (GeoSci., Weatherford, East Leake, Loughborough LE126JX, United Kingdom, said.assous@eu.weatherford.com), Laurie Linnett (none, Scotland, United Kingdom), and Peter Elkington (GeoSci., Weatherford, Loughborough, United Kingdom)

We consider the impact of processing method on the ability to differentiate between group and phase velocity in dipole sonic log data. A new array processing method is used to demonstrate that the choice of processing algorithm could lead to confusion. We emphasise the importance of knowing the limitations of the processing method used before interpreting results and applying them to the calculation of elastic rock properties. The proposed method addresses the confusion and clarifies that what is computed from the array waveforms can be either group or phase velocity depending on the algorithm used and the formation properties in the given situation. We demonstrate the effectiveness of the approach versus conventional processing methods on synthetic examples.

3:45–4:00 Panel Discussion

Session 2pUW

Underwater Acoustics: Reflection and Scattering from Ocean Surface and Bottom

Derek R. Olson, Chair
Department of Oceanography, Naval Postgraduate School, Monterey, CA 93943

Contributed Papers

1:30

2pUW1. Estimates of coherent reflection loss inclusive of the effects of near-surface bubbles on sound speed. Adrian D. Jones and Alex Zinoviev (Maritime Div., Defence Sci. and Technol. Group, P.O.Box 1500, Edinburgh, SA 5111, Australia, bearjones@adam.com.au)

Through simplification of the authors' model "JBZ" of coherent reflection loss at the ocean surface, a revised model has been obtained in extremely simple form. This model provides descriptions of the loss obtained at a wind-roughened surface, adhering to a Pierson-Moskowitz surface wave frequency spectrum, inclusive of the effects of near-surface bubbles on sound speed. A very compact form of the model has been obtained for application to surface ducted sound for a wide range of wind speed and frequency combinations, with the result that the dB loss with range is the same for all modes within the duct. The derivation of the model from the authors' earlier JBZ is described, and the conditions for equivalence are detailed. In a limited range of comparisons, descriptions of reflection loss per bounce obtained with the original JBZ and the compact version are shown to be nearly identical to corresponding values obtained by the "Relay" model described by Ainslie [*JASA* **118**, 3513–3523 (2005)].

1:45

2pUW2. Numerical investigation of the two-scale model for rough surface scattering. Derek R. Olson (Oceanogr., Naval Postgrad. School, 201 Appl. Sci. Bldg., University Park, Pennsylvania 16802, olson.derek.r@gmail.com)

The two-scale approximation for rough surface scattering assumes that a rough surface can be divided into two components: a large-scale and small-scale component, using a cutoff wavenumber to partition the wavenumber domain. This approach is attractive for multi-scale roughness such as the air-ocean and ocean-seafloor interfaces. Traditionally, the small roughness perturbation approximation is used for the small-scale component, and the Kirchhoff, or tangent-plane approximation is used for the large-scale component. Typically, the high-frequency approximation is used when propagating the surface fields to the far field, resulting in the physical interpretation of the large-scale surface modulating the local grazing angle of the small-scale surface. Using these approximations, the incoherent scattered power exhibits a dependence on the cutoff frequency, which is undesirable from a theoretical point of view. In this work, the validity of the approximations used on the small-scale surface, and large scale surface is examined through the use of numerical solution of the governing integral equations. This investigation is performed with the goal of testing the hypothesis that if each approximation on the component surfaces is demonstrated to be accurate, then the two-scale model should be independent of the cutoff frequency.

2:00

2pUW3. Time warping and acoustic characterization of the seafloor in horizontally inhomogeneous ocean. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu), Boris Katsnelson (Univ. of Haifa, Haifa, Israel), Tsu Wei Tan (Phys. Dept., Naval Postgrad. School, Monterey, CA), and Andrey Malykhin (Phys. Dept., Voronezh State Univ., Voronezh, Russian Federation)

Time-warping transform is increasingly employed in shallow water acoustics to separate the field due to a compact, broadband sound source or the two-point cross-correlation function of diffuse noise into their normal mode components, and to measure mode travel times as a function of frequency. The time-warping transform was developed for range-independent waveguides, while physical parameters of the ocean are never quite constant in the horizontal plane. Bathymetry variations are typically responsible for the bulk of the waveguide's range dependence as well as horizontal refraction of sound in the coastal ocean. Simple, exactly solvable models of shallow-water waveguides are used in this paper to illustrate the effects that the range dependence and horizontal refraction have on the performance of the warping transform and on the inferred geoacoustic parameters. Horizontal refraction due to generic bathymetric variations is addressed in the adiabatic approximation using perturbation techniques. Theoretical predictions are verified using numerical simulations. It is found that moderate bottom slopes can lead to large errors in retrieved geoacoustic parameters and cause positive bias in bottom sound speed estimates if horizontal refraction is ignored. [Work supported, in part, by NSF and BSF.]

2:15

2pUW4. Measurements of two-dimensional spatial coherence of normal-incidence seafloor scattering. Daniel C. Brown, Cale F. Brownstead (Penn State Univ., State College, PA 16804, dcb19@psu.edu), Anthony P. Lyons (Univ. of New Hampshire, Durham, NH), and Thomas B. Gabrielson (Penn State Univ., State College, PA)

The two-dimensional spatial coherence of the field backscattered from a complex lakebed has been characterized in a series of measurements made in Seneca Lake, New York. In the test region, the lakebed consists of a series of sediment layers created by a sequence of distinct depositional processes. The spatial coherence depends on the structure of the underlying sediment sequences. Significant ping-to-ping variability in the spatial coherence surface is observed for each sediment sequence. This variability is quantified by a two-dimensional spatial coherence metric that measures the coherence lengths and asymmetric coherence surface orientation. Sediment sequences with isotropic scattering strength exhibit random ping-to-ping variability in coherence length and coherence surface orientation. Sequences with spatially anisotropic scattering strength show intervals of non-random ping-to-ping variability in the coherence length and coherence surface orientation.

2:30

2pUW5. Sequential bottom parameter estimation using blind deconvolution of sources of opportunity in ocean waveguide for Santa Barbara channel experiment. Xuedong Zhang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Beijing 100190, China, zxd@mail.ioa.ac.cn), Nicholas C. Durofchalk, Karim G. Sabra (Georgian Inst. of Technol., Mech. Eng., Atlanta, GA), and Lixin Wu (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

This paper investigates the performance of sequential bottom parameter estimation based on ray-based blind deconvolution (RBD) [Sabra *et al.*, *JASA* EL42-7 (2010)] of sources of opportunity using the 2016 Santa Barbara Channel (SBC) experimental recordings of shipping noise. The RBD algorithm relies on estimating the unknown phase of the source of opportunity through wideband beamforming along a well-resolved ray path to approximate the environment's channel impulse responses (CIR) between the source and the VLA elements. The corrected power ratio of the direct and bottom-bounced arrivals is processed to infer the bottom reflection loss and is utilized to invert for the bottom parameters. Sequential parameter estimation uses a state space model for predicting and correcting the bottom parameters as the estimated bottom reflection loss values become available. Inversions results for the SBC experiment were also performed with conventional active sources to validate the inversion obtained with RBD of sources of opportunity.

2:45

2pUW6. Simulation and testing results for a sub-bottom imaging sonar. Daniel C. Brown, Shawn Johnson (Penn State Univ., State College, PA 16804, dcb19@psu.edu), Isaac Gerg (Penn State Univ., Port Matilda, PA), and Cale F. Brownstead (Penn State Univ., State College, PA)

The problem of detecting buried unexploded ordnance is addressed with a sensor deployed from a shallow-draft surface vessel. The sonar system produces three-dimensional synthetic aperture sonar imagery of both surficial and buried ordnance across a range of environments. The sensor's hardware design is based in part upon data created using a hybrid modeling approach that combined results from separate environmental scattering and target scattering models. This hybrid model produces synthetic sensor data where the sensor/environment/target space may be modified to explore the expected operating conditions. Based on these modeling results, a sonar system has been integrated to a test platform, and experiments have been conducted at a trial site in the Foster Joseph Sayers Reservoir near Howard, PA. Modeling and experimental results will be presented and discussed.

3:00

2pUW7. Modeling variability in spatial coherence measurements for seafloor scattering. Thomas E. Blanford, Daniel C. Brown, and Richard J. Meyer (Appl. Res. Lab., The Penn State Univ., The Penn State Univ., State College, PA 16804, teb217@psu.edu)

Correlation sonar systems, such as correlation velocity logs and synthetic aperture sonar micronavigation, can estimate platform motion using measurements of the spatial coherence of seafloor scattering. During the course of operation, the spatial coherence measurements made by these systems will fluctuate about their average values. A variety of factors, including noise sources, changing environmental properties, and statistical estimation error can all contribute to the variability in these measurements. This presentation will introduce models that can be used to predict the variability in measurements of spatial coherence. These models will be compared with field measurements of spatial coherence collected at normal incidence to the bottom of Seneca Lake. Finally, the application of these models will be discussed as they relate to the design of arrays and signal processing algorithms for correlation sonar systems. (The authors want to acknowledge the financial support for this work by Lockheed Martin Rotary and Mission Systems.)

3:15

2pUW8. Linear and half order fractional viscoelastic equivalents of the extended Biot model. Sri Nivas Chandrasekaran and Sverre Holm (Dept. of Informatics, Univ. of Oslo, Gaustadalléen 23B, Oslo 0373, Norway, srinc@ifi.uio.no)

The extended Biot poro-viscoelastic model [Chotiros and Isakson, *JASA* (2014)] like the standard Biot poroelastic model is biphasic and predicts two compressional waves (fast and slow) and one shear wave. Taking squirt flow into account, the model introduces two additional relaxation modes apart from the Biot crossover relaxation mode (global flow). The frequency dependent relaxation processes and a high number of independent parameters make the finite element implementation and the inversion of parameters for reconstruction a challenge. To overcome this challenge, we identify single phase viscoelastic equivalents for all three wave solutions of the extended Biot model from its dispersion relations. Considering squirt flow and high-frequency turbulent flow, the fast compressional wave and the shear wave solutions at low frequencies are equivalent to a non-standard solid model with eight parameters (four springs and four dashpots) and six parameters (Three springs and three dashpots), respectively. At high frequencies, the Kelvin-Voigt model used for tangential contact stiffness (shear relaxation) leads to non-physical equivalents. Neglecting this shear relaxation mode, the fast compressional wave and the shear wave solutions at high frequencies are equivalent to a fractional half-order solid model of order three and two, respectively. Alternative models for tangential contact stiffness are discussed.

3:30

2pUW9. Experimental study on measurement of transient sound in a reverberation tank. Rui Tang and Xinyue Yu (Harbin Eng. Univ., Nantong Str. Nangang Dist No.145, Harbin 150001, China, tangrui@hrbeu.edu.cn)

As a new application of reverberation tanks, the experimental study on measuring the sound energy level of transient sound sources in a reverberation tank was proposed in this paper. According to the law of energy conservation, the transient sound energy obtained by sound intensity integration in time domain, which is the same with sound energy density spectrum integration in frequency domain. By utilizing the invariant sound field correction factors, the transient sound source level could be obtained in a reverberation tank, and both the reverberant measurement methods in frequency domain and that in time domain were established. The impulsive sound and the spark sound experiments were carried out in UATL (Underwater Acoustics Technology Lab, Harbin Engineering University, China) to confirm the validity of the proposed method. Experiments results showed that the sound source level of the transient sounds could be precisely measured in reverberation tanks, and compared to free field measurement, the deviation was less than 1 dB.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

W. J. Murphy, Chair ASC S3
National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Cincinnati OH 45226

T. Ricketts, Vice Chair ASC S3
Vanderbilt University, 1215 21st Ave. South, Rm. 8310, Nashville TN 37232

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 14 May 2019.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance and comfort.

TUESDAY AFTERNOON, 14 MAY 2019 MORROW ROOM, 3:30 P.M. TO 4:45 P.M.

Meeting of Accredited Standards Committee (ASC) S3/SC 1, Animal Bioacoustics

D. S. Houser, Vice Chair ASC S3/SC 1
National Marine Mammal Foundation, 2240 Shelter Island Dr., Suite 200, San Diego, CA 92106

K. Fristrup, Vice Chair ASC S3/SC 1
National Park Service, Natural Sounds Program, 1201 Oakridge Dr., Suite 100, Fort Collins, CO 80525

Accredited Standards Committee S3/SC 1 on Animal Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43/SC 1 Noise and ISO/TC 43/SC 3, Underwater acoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 14 May 2019.

Scope of S3/SC 1: Standards, specifications, methods of measurement and test, instrumentation and terminology in the field of psychological and physiological acoustics, including aspects of general acoustics, which pertain to biological safety, tolerance and comfort of non-human animals, including both risk to individual animals and to the long-term viability of populations. Animals to be covered may potentially include commercially grown food animals; animals harvested for food in the wild; pets; laboratory animals; exotic species in zoos, oceanaria or aquariums; or free-ranging wild animals.

Meeting of Accredited Standards Committee (ASC) S12 Noise

S. J. Lind, Vice Chair, ASC S12
The Trane Co., 3600 Pammel Creek Rd., Bldg. 12-1, La Crosse, WI 54601 7599

D. F. Winker, Vice Chair, ASC S12
ETS-Lindgren Acoustic Systems, 1301 Arrow Point Dr., Cedar Park, TX 78613

Accredited Standards Committee S12 on Noise. Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1, Noise, and ISO/TC 43/SC 3, Underwater acoustics, take note - that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 14 May 2019.

Scope of S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control, including biological safety, tolerance and comfort, and physical acoustics as related to environmental and occupational noise.

TUESDAY EVENING, 14 MAY 2019 7:30 P.M. TO 9:00 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will meet starting at 4:30 p.m.

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 are as follows:

Engineering Acoustics (4:30 p.m.)	McCreary
Signal Processing in Acoustics (4:30 p.m.)	Beckham
Acoustical Oceanography	McCreary
Animal Bioacoustics	Clements
Architectural Acoustics	French
Musical Acoustics	Breathitt
Physical Acoustics	Jones
Psychological and Physiological Acoustics	Carroll Ford
Structural Acoustics and Vibration	Stopher

Committees meeting on Wednesday

Biomedical Acoustics	Nunn

Committees meeting on Thursday

Computational Acoustics (4:30 p.m.)	Clements
Noise	Segell
Speech Communication	Carroll Ford
Underwater Acoustics	McCreary