Session 2aAAa

Architectural Acoustics and ASA Committee on Standards: Acoustics of Lobbies, Atria, Stairways, Corridors, Pre-function, etc.

Logan D. Pippitt, Chair
Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045

Chair's Introduction—9:00

Invited Papers

9:05

2aAAa1. The face of the facility. Brandon Cudequest and K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd. Ste. 325, Westlake Village, CA 91362, bcudequest@gmail.com)

Lobbies and atria are a facility's initial destination, the point of departure, the information center, and the security checkpoint. They define a building's aesthetic, connect to the rest of the facility, require high speech intelligibility at key locations, and demand durability. The requirements can fluidly fluctuate throughout design, and the acoustics need to keep pace. This paper will discuss different projects where the lobby was central to a successful design. Highlights include a hospital atrium, wherein a balance was struck between speech intelligibility and speech privacy; an airport concourse where the prospect of tile grout lines was carefully considered; and a giant court-house lobby with design goals at odds with design standards, and with bridges providing acoustical "shade" for security guards.

9:30

2aAAa2. Acoustically neglected and ignored building spaces. Megan Stonestreet (Architecture, Univ. of Kansas, University of Kansas, Lawrence, KS 66045, meganstonestreet@ku.edu)

Often neglected and ignored building spaces in regard to room acoustics and building noise are corridors, stairways, and lobbies. For example, reverberant corridors in school teaching buildings are often noisy and provide an undesirable acoustic connection between classrooms; stairways are typically reverberant spaces that act as vertical corridors and are generally unpleasant spaces; lobbies are actually multipurpose spaces that serve as entrance, exit, and gathering spaces, and for performance facilities, they are also used as performance spaces. Examples of each of these building spaces will be discussed along with acoustic data and practical acoustic requirements.

9:55

2aAAa3. Case studies of two atria to begin a conversation regarding the importance of acoustical performance of transitional spaces. Michelle Huey and Michael (Mick) Barnhardt (D.L. Adams Assoc., 1536 Ogden St., Denver, CO 80218, mhuey@dlaa.com)

Transitional spaces, such as lobbies or atria, are largely forgotten spaces regarding room acoustics. These transitional spaces are widely used for presentations, meetings, and parties/gatherings. Lobbies and atria are often connected to hallways and adjacent office clusters, where sound can propagate to areas far away from the source. With the ever-increasing need for transitional spaces to also act as gathering spaces, it is important to begin a discussion of acoustical standards in transitional spaces. This paper discusses two case studies conducted by D.L. Adams Associates: (1) An atrium with a kitchen and dining area used for parties, gatherings, meetings, and throughout the day by employees; (2) an atrium with a commercial-grade kitchen at the center used for meetings, lunchtime, gatherings, and special events. This paper includes discussions of noise propagation, noise build up, reverberation time, and absorptive treatments within transitional spaces and how to make the spaces work acoustically, while isolating them from adjacent areas.

10:20

2aAAa4. Case study of acoustics in two very large atriums. Timothy Foulkes (Cavanaugh Tocci Assoc., 327 Boston Post Rd., Sudbury, MA 01776, tfoulkes@cavtocci.com)

Two recent projects have involved very large spaces with predominantly hard finishes. One assignment was to develop an acoustic treatment plan for an existing six story interior courtyard within an office tower. This space, a very formal design finished in marble, wood paneling, and glass, had a 7 second reverberation time. The courtyard was available as a lounge space to all tenants of the building but was barely utilized. We were part of a design team hired by the building owners to update the courtyard and make it more user friendly, including improvement in the acoustics. The other atrium was part of a new project—a seven story glass enclosure with a concrete floor. This atrium serves many different functions including the main entrance and security lobby, dining space, informal meeting space, and lounge. The acoustics were mitigated by the irregular shape, areas of carpet on the balconies, and limited areas of acoustic

paneling. The new atrium is a success and it has become the living room and hub for this large building. It is well used throughout the work day for all of the purposes mentioned above and more. Renovations for the other space are still in design as of the abstract deadline, but there may be finished in time for the Spring ASA meeting. The presentation will include a discussion of the dimensions and finishes, acoustic priorities, design challenges, acoustic measurements, and subjective comments.

10:45

2aAAa5. Measured acoustic conditions in the common spaces of two educational buildings at the University of Nebraska–Omaha. Samuel H. Underwood and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St. Omaha, NE, Omaha, NE 68182-0816, samuelunderwood@unomaha.edu)

This paper presents results from an acoustical case study of the atria, corridors, stairways, and other common spaces within two buildings located on the campus of the University of Nebraska-Omaha: the Peter Kiewit Institute and the Barbara Weitz Community Engagement Center. Both buildings, while educational in nature, differ drastically in acoustical design. Impulse responses, noise levels, and transmission loss in both facilities are measured and the resulting acoustic metrics are compared to recommended values for the respective spaces. Descriptions of each building's architectural features are presented and linked to the measurement results. In particular, many spaces in the Peter Kiewit Institute were designed to expose the building systems, and the lack of acoustical products such as acoustical ceiling tile results in higher noise levels than normally expected. The Weitz Center, however, was designed with great emphasis placed on achieving optimal indoor environmental conditions. The results of this case study support the importance of appropriate acoustic design atria, corridors, stairways, and other common spaces in educational settings.

11:10

2aAAa6. The architectural rejuvenation of Canada's National Arts Centre: The function of pre-function. Robin S. Glosemeyer Petrone and Marcus R. Mayell (Threshold Acoust.com, 53 W Jackson Blvd., Ste. 815, Chicago, IL 60604, robin@thresholdacoustics.com)

When Canada's National Arts Centre opened in 1967, the Brutalist architecture style of its era yielded a fortress for the arts. As Canada celebrates its sesquicentennial, a $110M (Canadian) architectural rejuvenation process is transforming the inward facing fortress into alluring beacon at the heart of the capital. Expanded public spaces enveloped in transparent facades form multiple pre-function spaces that visually spill into one another to convey the constant activity of Canada's incubator for the performing arts to the patrons within and the surrounding city. The desire for visual transparency, the building's designation as a National Historic Site of Canada, and need for simultaneous programming of the performance and pre-functions spaces provided no shortage of acoustic challenges.

Contributed Paper

11:35

2aAAa7. Creating realistic design goals through the use of locally-measured reverberation time and background noise data. Matt Whitney and Ted Pyper (K2, 5777 Central Ave., Ste. 225, Boulder, CO 80301, matt@k2avt.com)

One of the major difficulties in the design process of an atrium or lobby is providing points of reference to both the architect and the end user to help drive the conversation on an acceptable reverberation time and background noise level for the projected use cases. K2 recently surveyed eight different atrium and lobby spaces in Denver and used this reverberation time and background noise data to develop an appropriate range of reverberation times, for a given room volume, covering most of the common uses for atriums and lobbies. These ranges help provide guidance to both the design team and the end user on design goals and expectations for these spaces. This presentation will present these data, as well as subjective impressions and architectural design of each of the spaces, and describe how K2 used this information to initiate productive conversations with architects and end users and produce realistic design goals.

Session 2aAAb

Architectural Acoustics: Student Design Competition (Poster Session)

David S. Woolworth, Cochair
Roland, Woolworth & Associates, 356 CR 102, Oxford, MS 38655

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 and the National Council of Acoustical Consultants are sponsoring the 2015 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 US$1250 will be made to the submitter(s) of the design judged "first honors." Four awards of US$700 each will be made to the submitters of four entries judged "commendation."

TUESDAY MORNING, 8 MAY 2018 LAKESHORE B, 8:00 A.M. TO 11:55 A.M.

Session 2aAB

Animal Bioacoustics and ASA Committee on Standards: History of Animal Bioacoustics

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

Chair's Introduction—8:00

Invited Papers

8:05

2aAB1. A Brief history of avian bioacoustics. Robert Dooling (Psych., Univ. of Maryland, Baltimore Ave., College Park, MD 20742, rdooling@umd.edu) and Micheal L. Dent (Psych., Univ. of Buffalo, Buffalo, NY)

A proper history of avian bioacoustics would reference early naturalists such as Aristotle, Pliny the Elder, and others who observed that birds not only have acute senses but that they also learn their vocalizations with reference to auditory information and exhibit parallels with human speech, language, and music. On the production side, the development of the tape recorder followed by the sound spectrograph enabled researchers to precisely record, preserve, analyze, and quantify the characteristics of the amplitude and spectral envelope of the acoustic signals which birds use to communicate. In the 1950s Thorpe, and later Marler, tackled the development of vocal learning in birds and the nuances of individual and species differences, which began the modern era of avian bioacoustics. Almost simultaneously, on the perception side, the confluence of Skinnerian conditioning methods, signal detection theory, and the first laboratory minicomputers launched the modern era of animal psychophysics, which continues to this day in helping to understand the striking parallels between avian acoustic communication and speech and language learning in humans.

8:25

2aAB2. A history of fish bioacoustics. Arthur N. Popper (Univ. of Maryland, Biology/Psych. Bldg., College Park, MD 20742, apopper@umd.edu) and Anthony D. Hawkins (Loughine Ltd, Aberdeen, United Kingdom)

Awareness of fish sound production dates back to ancient times, and concern about effects of man-made sounds on fishes can be traced back at least to the mid 17th century. By the end of the 19th century, the morphology of the fish ear had been well described, but experimental studies of the hearing characteristics did not begin until the early 1900s, when Parker demonstrated that fish can detect

sounds. Subsequent work by von Frisch and his students, including Dijkgraaf, determined the range of frequencies that fishes could detect, and showed the fishes can discriminate between sounds. Later work in Europe provided a deeper understanding of sound detection mechanisms with Enger pioneering experiments on fishes in the sea. Experiments in the sea by Schuijf, Chapman, and others demonstrated that fish could discriminate sound directions. Meanwhile, in the U.S., Moulton, Tavolga, and their students documented diversity in fish hearing capabilities. Concurrently, studies by Fish, Winn, Myrberg, Tavolga, and others showed diversity in fish sounds, and documented the behaviors associated with fish sound production. Work by these investigators (and many others) opened the field of fish bioacoustics, and provided a wealth of information that stands today as the basis for modern studies.

8:45

2aAB3. Amphibian bioacoustics: From Arch to Zelick. Peter M. Narins (Integrative Biology & Physiol., UCLA, 621 Charles E. Young Dr. S., Los Angeles, CA 90095-1606, pnarins@ucla.edu)

Amphibian bioacoustics may be thought to have originated with the first published acoustic playback experiments with frogs 60 years ago. Martof and Thompson (1958, Behavior 13:243–258) and Littlejohn and Michaud (1959, Tex. Jour. Sci.11:86–92) showed that male calls of *Pseudacris nigrita* in Georgia are effective in attracting conspecific females, and that females of *Pseudacris streckeri* in Texas can discriminate conspecific calls from heterospecific calls, respectively. Capranica (1965, JASA 40:1131–1139) was the first to electronically synthesize bullfrog calls and to electronically modify (filter) them for use as stimuli in acoustic playback experiments. Since then, all hell broke loose. Reel-to-reel, cassette and DAT tape recorders, were replaced by wideband portable digital machines capable of recording directly to CF cards and/or to high capacity internal hard drives. Sophisticated call synthesis and analysis programs are widely available. Amphibian communication has been studied on six continents and Madagascar and we have gained a more sophisticated appreciation of the production and reception of amphibian airborne signals. Now the challenge is to find new and interesting amphibian bioacoustic questions, and for inspiration, I shall present a brief review of some of the noteworthy experiments of the past. [Work supported by NSF grant #1555734.]

9:05

2aAB4. History of technology for studying animal sound and communication. David K. Mellinger (Coop. Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu)

New technology has always been used to listen to, study, and understand animal sounds. Early recording devices in the 1870s employing wax cylinders and discs were quickly employed for recording nature sounds. Underwater listening devices from the 1910s captured marine animal sounds; interest in these sounds increased greatly after World War II, when naval advances made underwater sound accessible to sonar operators who noted the presence of cetacean sounds. Microphone technology advanced to the point where ultrasonic and eventually infrasonic signals were recorded. Frequency separation and analysis was facilitated by the invention of the vocoder in the late 1930s, which separated sound into a number of frequency bands and enabled, for instance, the discovery of multiple sources of sound (more than one syrinx) in bird vocalizations. The greatest leap in frequency separation came with the discovery of the Fast Fourier Transform and its application on digital computers for making spectrograms; since the 1970s, these simple graphic displays have made it possible to look at sound signals statically, greatly facilitating analysis. Advances in digital multi-channel recording have improved our ability to localize animals and study communication networks. Advances in analysis software continues to improve our ability to interpret sounds of animals.

Contributed Paper

9:25

2aAB5. A brief history of our understandings on underwater noise impacts to marine life and the evolution of its regulatory process in the United States. Shane Guan, Amy R. Scholik-Schlomer, and Jacqueline Pearson-Meyer (Office of Protected Resources, National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20910, shane.guan@noaa.gov)

It has been long understood that elevated noise is detrimental to signal detection by the anti-submarine warfare community. However, it was not until the early 1970s that similar concerns were raised regarding marine life that utilize sound for various life functions. The 1980s saw some of the first studies on effects of noise from offshore oil and gas exploration and development on marine mammals in Arctic waters. Low-frequency sources for ocean thermometry research and submarine detection over ocean basins brought additional concerns in the early 1990s on potential impacts on marine mammals from intense sources. Around late 1990s, several ad-hoc noise levels were adopted as acoustic impact thresholds of marine mammals by regulatory agencies in the U.S. Several cetacean mass stranding events that were coincidental with the mid-frequency military sonar operations and the increased awareness of ocean noise pollution accelerated research in this field in the 21st Century. Numerous studies on hearing sensitivity and noise induced temporary threshold shift or physical injuries on various marine animal taxa led to new sophisticated regulatory guidelines on assessing underwater noise impacts to marine life. In addition, recent understanding of soundscapes as environmental quality factors provides new perspectives on marine species and ecosystem conservation.

9:40–9:55 Break

Invited Papers

9:55

2aAB6. The Office of Naval Research Marine Mammal Program, 1990–2006. Robert Gisiner (IAGC, 5410 Sycamore Villas Dr., Kingwood, TX 77345, gisiner@aol.com)

ONR first stood up a dedicated marine mammal research program in 1990. The initial effort was led by Dan Costa, with the support of experienced ONR program leaders like Mel Briscoe, Bernie Zahuranec, and Steve Zornetzer. The author was privileged to manage the program from 1993 to 2006; a period of rapid expansion of concern about the marine environment accompanied by the desire to put tools for improved understanding in the hands of scientists and decision makers. The emphasis from the start was on multi-disciplinary collaborations. Major research themes included hearing and exploration of Temporary Threshold Shift as a metric of auditory risk; advancement of animal-borne scientific instruments ("tags"); advancement of ocean acoustic monitoring technologies, refinement of field sound exposure experimental methods, and improvement of sound exposure models. Frequent external reviews, including four National Research Council (NRC) reviews, were vital to the quality and credibility of the program. The 2005 NRC review on the Population Consequences of Acoustic Disturbance (PCAD) is a textbook example of how ONR's basic research contributions have had far-reaching consequences that greatly exceeded initial expectations. The convergence of growing concern about manmade sound in the ocean and the ONR model for quality in basic research has had a profound impact on our ability to understand ocean bioacoustics and to make better informed decisions about the potential risks posed by human activities in the marine environment.

10:15

2aAB7. The last decade of bioacoustics at the Office of Naval Research. Michael Weise (Office of Naval Res., 875 N Randolph Sr, Arlington, VA 94598, michael.j.weise@navy.mil)

The goal of the ONR Marine Mammals and Biology (MMB) program is to enable Navy to and meet operational training and testing objectives in an environmentally responsible and legal manner. The ONR Marine Mammals and Biology program invests in Monitoring and Detection topic with the goal is to improve marine mammal monitoring capabilities over current methods. Develop new and existing technology such as passive acoustics, IR, and others. Recent research efforts on passive acoustics include the development and testing of new autonomous hardware platforms and signal processing algorithms for detection, classification, and localization of marine mammals. Ultimately, the ONR goal is to adapt those algorithms for use on a variety of fixed, towed, floating, and profiling platforms. For example, over the last several years ONR has adapted the use of autonomous ocean gliders for marine mammal monitoring to create the desired capability of persistent, autonomous, passive acoustic monitoring of an area for marine mammal presence and abundance to provide timely, reliable, accurate, and actionable information to support marine mammal mitigation and monitoring. A key goal of ONR sponsored technology development is making the technology available to the broader Navy and research communities.

10:35

2aAB8. The history of marine mammal bioacoustics at the Navy Marine Mammal Program. Dorian S. Houser (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org), James J. Finneran, Keith Jenkins, E. E. Henderson (US Navy Marine Mammal Program, San Diego, CA), Jason Mulsow, Brian K. Branstetter (National Marine Mammal Foundation, San Diego, CA), Patrick Moore (National Marine Mammal Foundation, Spring Valley, CA), and Stephen W. Martin (National Marine Mammal Foundation, San Diego, CA)

The United States Navy Marine Mammal Program (MMP) has been in existence for over 50 years. Following its inception, the program quickly became involved in the study of marine mammal sensory systems and bioacoustics. Early studies included the pioneering work of C. Scott Johnson in obtaining the first behavioral audiogram in a dolphin and Sam Ridgway's electrophysiological studies of dolphin hearing and sound production. Marine mammal bioacoustic studies grew substantially in the decades following the MMP's inception, and included numerous investigations into odontocete biosonar, pinniped and odontocete hearing (using both behavioral and physiological methods), and the impact of human-made sound on the hearing, behavior, and physiology of marine mammals. The MMP's bioacoustic research has significantly contributed to the Navy's environmental stewardship mandate (i.e. to predict and mitigate the impact of Navy activities on marine mammals), the development of bio-inspired sonar systems, and the ability to assess the hearing capabilities of marine mammals in the wild, under human care, and in stranded or rehabilitation scenarios. The MMP continues its bioacoustic studies today with investments focused on bottlenose dolphin and sea lion bioacoustics, but with expansion to the passive acoustic monitoring of wild marine mammals.

10:55

2aAB9. History of marine bioacoustics in Florida and the Caribbean. David Mann (Loggerhead Instruments, Inc., 6576 Palmer Park Circle, Sarasota, FL 34238, dmann@loggerhead.com)

The diverse fauna of Florida and the Caribbean has long attracted scientists studying marine bioacoustics. The earliest studies (1905–1945) in Florida examined species distributions and life histories. After World War II bioacoustics studies began in earnest. At Marineland, which opened to support underwater filming, Lilly studied dolphin vocalizations and attempted to link them to behavior. Melba and David Caldwell pioneered studies on the development of dolphin signature whistles. Tavolga performed fundamental work on sound production and behavior in gobies and many studies on the hearing ability of fishes at Bimini, Marineland, and Mote Marine Laboratory. Myrberg's Miami laboratory showed that different damselfishes had distinct sounds. In the 1960's, Breder documented diel and seasonal variation in fish sound production. Most recently, studies have focused on wild animals. Working with Wells' Sarasota Dolphin Research Project, Tyack, Sayigh, and Janik have cataloged signature whistles and used playback experiments to study salient features. Fundamental work identifying sounds produced by fishes begun by Lobel has spread to several research groups in Florida, USVI, Puerto Rico, and the Cayman Islands. Passive acoustic studies using gliders show that much work remains to identify the sources of unknown sounds.

11:15

2aAB10. History of bioacoustics research on aquatic and marine organisms in Hawaii. Whitlow Au (Univ. of Hawaii, P.O. Box 1346, Kaneohe, HI 96744, Kailua, HI 96734, wau@hawaii.edu)

Modern bioacoustic research with aquatic animals began in Hawaii in the 1968–1969 period when three independent programs started almost simultaneously. Two faculty members began studying various facets of hearing in fish at the University of Hawaii. Another faculty member from a different department established a laboratory in which dolphin hearing and echolocation were among some of the topics studied. The Navy started a dolphin facility to train dolphins to perform Navy related tasks. The Navy program also included research on dolphin echolocation including a study comparing the performance of two dolphins and a Straza-500 CTFM sonar in target detection. The Navy facility closed in 1993 but some of the bioacoustics research was transferred to the University's Hawaii Institute of Marine Biology on Coconut Island about a mile from the former Navy facility in the same bay. The HIMB program expanded into field research with spinner dolphins and humpback whales, and passive acoustic monitoring of marine organisms. The local National Marine Fisheries Service of NOAA became a close partner in the PAM development and studies. This presentation will focus of the bioacoustic research performed with marine mammals highlighting some of the more important findings and contributions to the field.

11:35

2aAB11. The central role of Woods Hole Oceanographic Institution (WHOI) in marine mammal bioacoustics. Douglas Wartzok (Dept. of Biological Sci., Florida Int. Univ., 11200 SW 8th St., Miami, FL 33199, wartzok@fiu.edu) and G. Carleton Ray (Environ. Sci., Univ. of Virginia, Charlottesville, VA)

WHOI researchers pioneered marine mammal bioacoustics. Schevill and Lawrence in 1949 made the first recordings of marine mammals, beluga whales, and in 1956 first demonstrated cetacean echolocation. Schevill later concentrated on the taxonomy and behavior of cetaceans while Watkins developed the first portable high-frequency recorder and passive and active acoustic tracking systems. Ray, Watkins, and Schevill in 1969 presented the first evidence of song in a marine mammal, the bearded seal, linked to the behavior of males in the breeding season. Watkins' papers on vocalizations informed a generation of bioacousticians on how to interpret sonograms. He was one of the first to use the Navy's SOSUS hydrophone array to track cetacean movements. Schevill and Watkins also inspired the research of others, both in recoding marine mammal sounds and in interpreting sound in behavior. Tyack led the development of the D-Tag, which simultaneously records vocalizations and movements of cetaceans. Ketten used CAT scans to develop hearing models for cetaceans. Sayigh advanced knowledge of delphnid signature whistles and Fristrup further developed the concept of soundscapes. Watkins and Schevill's database of more than 20,000 vocalizations from 70 marine mammal species now resides at the New Bedford Whaling Museum, freely available to the public.

TUESDAY MORNING, 8 MAY 2018 GREENWAY A, 8:00 A.M. TO 10:35 A.M.

Session 2aAOa

Acoustical Oceanography and Underwater Acoustics: Acoustic Seabed Characterization I

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

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

Invited Papers

8:00

2aAOa1. Bayesian geoacoustic inversion for sediment profiles consisting of a general gradient over a layered substrate. Stan E. Dosso, Josée Belcourt (School of Earth & Ocean Sci, Univ of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca), Jan Dettmer (GeoSci., Univ. of Calgary, Calgary, AB, Canada), and Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA)

This paper considers Bayesian geoacoustic inversion and uncertainty estimation for seabed profiles consisting of a smooth gradient of general form overlying a potentially-layered substrate. The profile gradient is parameterized as a linear combination of Bernstein-polynomial basis functions, with the polynomial order determined objectively by applying the deviance information criterion. In this way the form of the gradient is determined by the data, rather than by a subjective prior choice. The substrate below the gradient consists

of uniform layers defined by an unknown number of (zero or more) interfaces, sampled probabilistically with trans-dimensional inversion. (Zero interfaces corresponds to a half-space with uniform parameter values equivalent to those at the base of the gradient.) The inversion approach is well suited to seabed environments consisting of an upper layer of soft, fine-grained sediments in which geoacoustic properties vary as a smooth function of depth overlying a harder, layered substrate, such as at the ONR Seabed Characterization Experiment at the New England mud patch. [Research funded by the Office of Naval Research, Ocean Acoustics Program.]

8:20

2aAOa2. Geoacoustic inversion of the New England Mud Patch using modal time-frequency dispersion. Julien Bonnel (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu), Dimitrios Eleftherakis (LabSTICC, ENSTA Bretagne, Brest, France), Stan E. Dosso, N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), David R. Dall'Osto, and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

This paper presents geoacoustic inversion of several low-frequency broadband signals recorded during the Seabed Characterization Experiment (SBCEX) that took place on the New England Mud Patch in March 2017. The considered sources are chirps emitted by a towed J15 source and underwater impulses created by MK64 explosions. These sources are low-frequency ($f<250$ Hz) so that the shallow water environment acts as a dispersive waveguide, and propagation is conveniently described by modal theory. The received signal is recorded on a single hydrophone placed 0.8 m off the bottom. In this context, inversion is carried out by matching modal dispersion curves in the time-frequency domain. Experimental dispersion curves are estimated using a non-linear sampling scheme called warping. Up to six modes can be estimated, and non-linear inversion results are consistent with what is known about the area.

8:40

2aAOa3. Inversions for sediment parameters using geophone and tetrahedral array measurements in the New England Mud Patch. Gopu R. Potty, James H Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), Ying-Tsong Lin, Julien Bonnel, and Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Measurements of acoustic pressure and particle velocity were made during the Seabed Characterization Experiment (SCEx) in the New England Mud Patch south of Cape Cod in about 70 meters of water. The University of Rhode Island and Wood Hole Oceanographic Institution deployed the "geosled" with a four-element geophone array, a tetrahedral array of four hydrophones, and several hydrophone receive units (SHRUs). In addition, a new low frequency source, interface Wave Sediment Profiler (iWaSP) was deployed to excite interface waves (Scholte waves). The iWaSP system consists of a source to generate the interface wave and a four-element accelerometer receive array. Results of inversions for geoacoustic parameters will be presented. Arrival time dispersion from broadband sources (SUS and CSS) and iWaSP will be used for these inversions. Both p-wave and Scholte wave arrivals will be investigated using the geophone and hydrophone data. Seismic data collected at the location will be used to constrain the model parameters in the inversion. Sediment data from cores, other *in-situ* measurements and inversions using other types of data will be used to compare and validate our inversions. [Work supported by Office of Naval Research.]

9:00

2aAOa4. Attenuation in fine-grained sediments for New England Mudpatch in 25–5000 Hz band. David P. Knobles (KSA, LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and William S. Hodgkiss (UC San Diego, La Jolla, CA)

A subject of considerable importance in littoral ocean environments is the frequency dependence of sediment attenuation for various classes of sediments. The well-known frequency exponent value of 1.8 for sandy sediments was the subject of numerous research studies. For mud-like or mixed sediments, for which there is a scarcity of experimental data, values for the frequency exponent are not well established. Previously, the sound speed structure of a surface mud layer in the New England Seabed Characterization experimental area was estimated with inversion and Bayesian methods. The inversions utilized MK-64 SUS explosive sources in a 25-275 Hz band placed on concentric circles of 2, 4, and 6.5 km radii in the latitude-longitude plane. The area was previously surveyed with CHIRP sonar allowing for multiple sediment layer horizons to be constrained for layer thicknesses by measured two-way travel times. In this analysis, a larger bandwidth (25-5000 Hz) of the SUS, the Combustive Sound Source (CSS), and tonals from towed sources at multiple frequencies allow for the development of posterior probability distributions that contain the statistics of the frequency dependence of sediment attenuation up to about 5 kHz. [Work supported by ONR]

9:20–9:35 Break

Contributed Papers

9:35

2aAOa5. Source characteristics and variability of impulsive sources used in ocean acoustics experiments. Matthew C. Zeh, Preston S. Wilson, Andrew R. McNeese (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 3374 Lake Austin Blvd., Apt. B, Austin, TX 78703, mzeh@utexas.edu), and David P. Knobles (Knobles Sci. and Anal., LLC, Austin, TX)

Two types of impulsive sources were used extensively in the Seabed Characterization Experiment 2017 (SCE2017): Mk-64 Sound Underwater Source (SUS), and the Combustive Sound Source (CSS). Both of these sources rely on the release of chemical energy and hence the radiated acoustic signal is subject to variability from shot to shot. In addition, SUS activates via a pressure sensor and there is variability in the depth of activation, which translates into additional variability in the radiated acoustic signal. A pair of ship-deployed hydrophones was used during SCE2017 to record source waveforms, and in this presentation, various acoustic characteristics of these sources, such as source level, frequency content, and the nature of the time domain signals will be described in terms of their statistical variability. [Work supported by ONR.]

9:50

2aAOa6. Distinctions among physical types of marine sediments, especially mud, and their possible fundamentals-based and heuristic models. Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann, and Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Marine sediments at different locations and depths are not necessarily similar. Distinctions are (1) sandy/silty sediments, when most particles have settled and almost every particle touches other particles, and (2) mud sediments, where the majority in a layer are suspended so that they do not touch other particles. Most theoretical and analytical modeling over the years has largely been confined to sandy/silty sediments, and some successful models, although arguably heuristic and with adjustable parameters, have emerged. There is also some speculation that such models might apply equally well for mud sediments. The present authors in recent years have concentrated on the development of a fundamentals-based model for mud sediments, although there is as yet insufficient data to adequately test its validity. The present paper speculates on what this model may have in common with models for sandy/silty sediments. There are expectations that the Mallock-Wood model may approximately apply for sound speed predictions at very low frequencies. Arguments are presented to the effect that the physics is sufficiently different for the two sediments types that a substantially different model is needed for mud sediments. One chief distinction is that the solid portion of mud sediments has a portion that is clay particles, and these have substantially different electrolytic and cohesive properties than those of quartz particles.

10:05

2aAOa7. Acoustics of biologically active marine sediments. Kevin M. Lee, Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Gabriel R. Venegas, and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Biological organisms are prevalent on and near the seabed; however, most predictive theoretical models do not explicitly address the acoustic behavior of biologically active marine sediments. Sediment-acoustics models, such as those based on Biot theory or grain-shearing theory, allow for parameterization of various physical properties including bulk density, porosity, shear strength, or pore fluid viscosity, all of which can be modified by the presence of biological organisms or bioturbation. Other models can include the effects gas bubble distributions, which can also be associated with the presence of benthic biology, on compressional and shear wave propagation. In this paper, we examine sediment-acoustics models in comparison with data from various field experiments focused on infauna- and seagrass-bearing sediments. The model parameters are partially constrained by physical data from core samples collected at the field experiment sites. Empirical relations between the core data parameters and acoustic data will also be examined. [Work supported by ONR and ARL:UT IR&D.]

10:20

2aAOa8. Implications from a clay/silt suspension model of mud for data from the Seabed Characterization Experiment. Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Rensselaer Polytechnic Inst., Dept Mathematical Sci., Troy, NY 12180, browne6@rpi.edu), Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., East Sandwich, MA), Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Jason D. Chaytor (Coastal and Marine Sci. Ctr., U. S. Geological Survey, Woods Hole, MA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Mud-like marine sediments may be modeled as a suspension of flocculated clay particles in water, in which silt particles are embedded in the clay flocs. Recent calculations based on this model [Pierce *et al.*, New Orleans ASA] produced new predictions for the frequency-dependent phase speed and attenuation of compressional waves. For phase speed, very good comparisons between model predictions and archival data were found. Comparisons will be presented using geoacoustic data from sediment cores extracted from the New England Mud Patch. For attenuation, predictions using Gaussian distributions of silt particle radius were calculated. An approach will be described for producing a frequency-dependent effective particle radius to account for size-distribution effects using single-particle attenuation expressions. The influence of non-Gaussian size distributions on the effective radius and on the frequency intervals of near-linear attenuation behavior will be shown. An environmental model for a WHOI AUV track deployed in the ONR Seabed Characterization Experiment was constructed, and initial comparisons were made between transmission loss calculations and acoustic data. Benchmarking will be performed with additional forward calculations, along with estimates of geoacoustic properties. Particular interest is in sediment attenuation and its frequency dependence. [Work supported by ONR.]

Session 2aAOb

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

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

Chair's Introduction—11:00

Invited Paper

11:05

2aAOb1. Three-dimensional shallow water sound propagation and applications toward acoustical oceanography. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

Underwater sound propagation in areas of the continental shelf, shelf break, and continental slope can encounter strong horizontal reflection, refraction, diffraction, focusing, and/or defocusing due to a variety of environmental factors, including bathymetric variability, water column fluctuations (internal waves and shelf-break fronts), and boundary roughness. Theoretical, numerical, and experimental approaches have been taken to investigate the underlying physics of these three-dimensional (3-D) sound propagation effects and their temporal and spatial variability induced jointly by marine geological features and dynamic oceanographic processes. Some recent work on theoretical analysis, numerical modeling and field work experiments to study 3-D sound propagation in nonlinear internal wave ducts, over the continental slope and in submarine canyons will be reviewed in this talk. The ultimate goal of investing these 3-D sound propagation effects is to improve long-distance acoustical oceanographic technology in complex ocean environments. An example using a 3-D back-propagation method for acoustic inverse problems will be presented, along with discussion on feature applications. [Work supported by the Office of Naval Research.]

Session 2aBA

Biomedical Acoustics: Using Acoustic Wave Propagation to Estimate Quantitative Material Properties of Tissue I

Matthew W. Urban, Chair
Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905

Chair's Introduction—8:00

Invited Papers

8:05

2aBA1. Magnetic resonance elastography: Status of clinical applications. Richard L. Ehman (Dept. of Radiology, Mayo Clinic, 200, Rochester, MN 55905, ehman.richard@mayo.edu)

Magnetic resonance elastography (MRE) is an imaging technology that has been available as an FDA-cleared upgrade to MRI systems since 2009. Worldwide, MRE technology has been installed on approximately 1000 MRI systems. In October 2017, the AMA approved the creation of a CPT code for MRE. In MRE, propagating shear waves in the range of 30–300 Hz are generated in tissue, and imaged with a phase contrast MRI technique. The wave images are processed with an inversion algorithm to generate cross-sectional images quantitatively depicting the complex shear modulus of tissue. The main application of MRE is currently in noninvasive diagnosis of chronic liver disease. Published studies have established the MRE is the most accurate and technically successful non-invasive modality for diagnosing liver fibrosis. Many other promising applications of MRE are being explored by investigators around the world, including assessment of diseases of the brain, heart, lung, and muscle and tumors of breast, liver, prostate, liver, and other organs. The new applications that closest adoption in clinical practice are (1) assessment of brain disease and (2) advanced diagnosis of liver disease. These new applications require acquisition sequences and processing algorithms that are more advanced than the current product MRE techniques.

8:25

2aBA2. Magnetic resonance elastography inversions: Technical challenges and recent developments. Armando Manduca (Physiol. and Biomedical Eng., Mayo Clinic, Opus 2-125, Mayo Clinic, 200 1st St. SW, Rochester, MN 55901, manduca@mayo.edu)

Magnetic resonance elastography (MRE) is a phase contrast based MRI imaging technique that can quantitatively and non-invasively measure full 3D vector displacement data from propagating acoustic waves *in vivo*. The data acquired allow the calculation of local values of complex shear modulus and the generation of images that depict the viscoelastic properties of tissue. Acquisitions at different frequencies can capture the dispersive behavior of these quantities, and advanced approaches attempt to map anisotropic and poroelastic properties of materials. We will discuss current approaches for the inversion of MRE data, focusing on the issues and assumptions involved, and highlight some of the challenges faced in such inversions, particularly in thin or small structures in which wave propagation is dominated by waveguide effects. We will also present results based on neural network inversions, which may have lower repeatability error and be more resistant to noise than some current approaches.

8:45

2aBA3. Spectral-based quantitative ultrasound imaging: A model free approach. Michael L. Oelze, Trong Nguyen, and Minh Do (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Mathews, Urbana, IL 61801, oelze@uiuc.edu)

Quantitative ultrasound (QUS) imaging can improve the diagnostic capabilities of ultrasound. Spectral-based QUS approaches utilize backscattered ultrasound signals to extract additional information about tissue state. These technique have traditionally relied on scattering models to differentiate tissue state. However, the use of machine learning approaches may obviate the need for models. A model-based and model-free (principle component analysis (PCA)) approach to spectral-based QUS were compared for their ability to differentiate fatty from non-fatty liver in a rabbit model. PCA was observed to provide better differentiation of fatty from non-fatty liver, i.e., PCA predicted fatty liver 86% of the time versus 36% for model-based. In addition, three calibration approaches for spectral-based QUS were compared: the traditional reference phantom, reference free, and an *in situ* calibration target. To test the calibration procedures, a phantom was scanned with and without a lossy layer placed on top and integrated backscatter coefficients (IBSCs) were calculated from the scattered data. Utilizing an *in situ* calibration target provided the ability to account for transmission losses and attenuation. The root mean square error between IBSCs estimated from the phantom with and without the lossy layer were 8.28 for the reference phantom versus 1.24 for the *in situ* calibration approach.

9:05

2aBA4. Fast simulations of shear waves generated by an acoustic radiation force: Comparisons with measurements in shear wave phantoms. Yiqun Yang (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI), Matthew W. Urban (Dept. of Radiology, Mayo Clinic College of Medicine and Sci., Rochester, MN), and Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI, mcgough@msu.edu)

We recently developed a fast approach for simulating shear waves generated by an acoustic radiation force. The input for these simulations is generated in FOCUS, the "Fast object-oriented C++ simulator" (www.egr.msu.edu/~fultras-web), which rapidly models the three-dimensional (3D) pressure, intensity, and acoustic radiation force on a desktop computer. The shear waves generated by the acoustic radiation force are then quickly simulated with Green's functions for viscoelastic media in a two-dimensional (2D) plane on a high-performance graphics processing unit (GPU). For these simulations, an L7-4 linear array is electronically focused at a depth of 25 mm to generate an acoustic radiation force "push" for 200 microseconds. The simulation results are compared to shear wave data measured in three viscoelastic shear wave phantoms with low, medium, and high values of the shear viscosity using a Verasonics Vantage ultrasound system. The measured and simulated shear wave data are compared for several different combinations of the simulation parameters, where these comparisons are enabled by the rapid acoustic radiation force and shear wave simulations. The results show that the shear waves simulated in 2D planes achieve good agreement with the measured shear wave data. [Work supported in part by NIH Grants EB023051 and DK092255.]

9:25

2aBA5. Efficient computational algorithms for modeling guided wave dispersion in arterial walls. Ali Vaziri (MSC Software Corp., Raleigh, North Carolina), Matthew W. Urban (Mayo Clinic, Rochester, MN), Wilkins Aquino (Duke Univ., Durham, NC), James F. Greenleaf (Mayo Clinic, Rochester, MN), and Murthy Guddati (NC State Univ., 2501 Stinson Dr., NCSU-Civil Eng., Raleigh, NC 27695-7908, mnguddat@ncsu.edu)

Arterial stiffness is a well-known biomarker of early cardiovascular disease. Shear wave dispersion ultrasound vibrometry (SDUV) has emerged as a promising technique to estimate local arterial stiffness from the observed dispersion of guided waves. With the ultimate goal of developing real-time inversion for arterial stiffness from SDUV measurements, we develop and validate highly efficient and accurate computational algorithms that compute the wave dispersion in multi-layered immersed tubes. The proposed approach carefully combines Fourier transformation and *one-dimensional* finite-element discretization to accurately capture fully *three-dimensional* wave propagation. The method is several orders of magnitude more efficient than three-dimensional finite element simulation, and eliminates other complexities such as the need for absorbing boundary conditions. The method is validated using SDUV experiments on tissue-mimicking phantoms. The validation exercise uncovered an important detail that is often overlooked—the dispersion curve captured through SDUV experiments does not correspond to a single dispersion curve, but a combination of multiple dispersion curves. This implies that proper identification of the dispersion curves could be critical to estimating the arterial stiffness. In this talk, we present the details of the proposed method including formulation, computational complexity, verification and validation.

9:45

2aBA6. Speckle-free estimation of tissue elasticity with single track location shear wave elasticity imaging. Peter J. Hollender (Biomedical Eng., Duke Univ., Rm. 1427, FCIEMAS Box 90281, 101 Sci. Dr., Durham, NC 27708, peter.hollender@duke.edu)

Shear Wave Elasticity Imaging (SWEI) is commonly used to characterize tissue elasticity, but conventional, multiple-track-location SWEI (MTL-SWEI) techniques are resolution-limited by speckle. MTL-SWEI techniques use plane wave ultrasound to monitor an induced shear wave as it propagates across a set of tracking beams within a region of interest. The scattering process creates a random, yet stationary, spatial sensitivity pattern for each beamformed location, called speckle bias, which causes errors in MTL-SWEI shear wave speed estimates that cannot be improved through averaging. Single Track Location SWEI (STL-SWEI) overcomes speckle bias by using a single track beam, subsequently exciting and tracking different push locations, and comparing the timing of the recorded shear wave signals from different push locations to estimate shear wave speed. Two and three-dimensional STL-SWEI imaging techniques are presented and compared to MTL-SWEI and Acoustic Radiation Force Impulse (ARFI) imaging in phantoms and *in vivo* experiments. Tradeoffs and techniques for sequencing, beamforming, shear wave speed estimation, and image formation are discussed. For applications where tissue heating and motion are not limiting factors, STL-SWEI provides superior imaging in terms of lateral resolution and contrast-to-noise ratio compared to MTL-SWEI and ARFI. [This work was supported by NIHR37HL096023 and NIHR01EB01248.]

10:05–10:20 Break

10:20

2aBA7. Identifying nonalcoholic fatty liver disease from quantitative ultrasound measures. W. D. O'Brien, Jr., Aiguo Han (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 306 N. Wright St., Urbana, IL 61801, wdo@uiuc.edu), Ethan Z. Sy (Dept. of Radiology, Univ. of California at San Diego, La Jolla, CA), Rohit Loomba (Dept. of Medicine, Univ. of California at San Diego, La Jolla, CA), Claude B. Sirlin (Dept. of Radiology, Univ. of California at San Diego, La Jolla, CA), John W. Erdman (Dept. of Food Sci. and Human Nutrition, Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Michael P. Andre (Dept. of Radiology and the San Diego VA Healthcare System, Univ. of California at San Diego, La Jolla, CA)

Nonalcoholic fatty liver disease (NAFLD) is the most common cause of chronic liver disease in the United States, affects 30% of adult Americans, may progress to nonalcoholic steatohepatitis (NASH) and end-stage liver disease, and is a risk factor for diabetes and cardiovascular disease. The diagnosis, grading, and staging of NAFLD currently is based on liver biopsy with histologic analysis. Noninvasive image-based methods to evaluate the liver in adults with NAFLD are urgently needed. The objective is to identify the relationships between quantitative ultrasound (QUS) outcomes [backscatter coefficient, BSC, and attenuation coefficient, AC] and tissue properties to validate the diagnosis and/or grading of NAFLD. 83 participants with known or suspected NAFLD (thus few uncompromised liver samples expected) received contemporary QUS and MRI (proton density fat fraction) PDFF to estimate liver fat fraction. Of this group, 74 participants also received liver biopsy. Participant recruitment is continuing. Currently, observations show good correlations of both BSC and AC with PDFF, where PDFF ranged from 3% to 42%. Of the 74 biopsy participants, all stages of fibrosis have been pathologically identified [F0,25; F1,22; F2,13; F3,10; F4,4] suggesting both QUS identification opportunities and confounder challenges. [Support: NIH R01DK106419 and NIH R37EB002641.]

10:35

2aBA8. A nonlinear grid-search inversion for cortical bone thickness and ultrasonic velocities. Tho N. Tran (Dept. of Radiology and Diagnostic Imaging, Univ. of AB, Res. Transition Facility, Edmonton, AB T6G 2V2, Canada, tho.tran@ualberta.ca), Mauricio D. Sacchi (Dept. of Phys., Univ. of AB, Edmonton, AB, Canada), Vu-Hieu Nguyen (Laboratoire Modélisation et Simulation Multi Echelle UMR 8208 CNRS, Université Paris-Est, Creteil, France), Dean Ta (Dept. of Electron. Eng., Fudan Univ., Shanghai, China), and Lawrence H. Le (Dept. of Radiology and Diagnostic Imaging, Univ. of AB, Edmonton, AB, Canada)

Cortical thickness and elasticity are important determinants of bone strength. This study proposes a nonlinear grid-search inversion to estimate the thickness and ultrasonic velocities of long cortical bones from axially-transmitted data. The inversion scheme has been developed in the frequency-phase velocity domain to recover bone properties. The method uses ultrasonic guided waves to retrieve cortical thickness, compressional, and shear-wave velocities of the cortex. The inversion strategy requires to systematically examine a set of trial dispersion curves within a pre-defined model space to match the data which minimizes the objective function in a least-squares sense. The theoretical dispersion curves required by the inversion are computed for bilayered bone models using a semi-analytical finite-element method. Its application demonstrates the feasibility of the proposed approach on synthetic data for a 5 mm-thick bone plate, and *in-vitro* dataset from a bovine femur plate with 2 mm-thick soft-tissue mimic on top. Our results indicate that one can recover the cortical thickness and wave speeds with less than 5% error. The accuracy can be improved by refining the grid; however, the computational cost of the method will increase. In the future, we aim at applying the method to clinical data.

10:50

2aBA9. Characterization of ultrasound attenuation and angular distribution from trabecular bone phantom material using refracto-vibrometry. Benjamin A. Rorem, Matthew R. Mehrkens, Thomas M. Huber (Phys., Gustavus Adolphus College, 800 W College Ave., Saint Peter, MN 56082, huber@gac.edu), Matthew T. Huber, and Brent Hoffmeister (Phys., Rhodes College, Memphis, TN)

Ultrasound is being researched and used for diagnosing osteoporosis. Researchers use trabecular bone from humans as a test material, but such bone is bio-hazardous, not uniform, and of limited size. Recently it has been demonstrated that the open-cell polyurethane foam, known as "Sawbones" could be utilized as a substitute for trabecular bone in ultrasound studies. In the current study, rectangular slices of Sawbones trabecular bone phantom material were insonified with a 25 mm diameter 500 kHz ultrasound transducer oriented both normal to the surface and at an angle. Refracto-vibrometry (RV), an interferometric method for optically measuring ultrasound, was compared with conventional transducer measurements of ultrasound transmission through the bone phantom samples. The measurement beam from a Polytec PSV-400 scanning laser Doppler vibrometer was directed through a water tank towards a stationary retroreflective surface. Acoustic wave fronts (density variations) which pass through the ~50μm diameter measurement laser cause variations in the integrated optical path length. The measured signals were used to determine parameters such as normalized broadband ultrasonic attenuation (nBUA) at numerous scan points. This enabled measurements of the spatial and angular distribution of the transmitted ultrasonic field through the sample that are not possible using a conventional single-element ultrasonic transducer.

11:05

2aBA10. Characterizing anisotropic, viscoelastic material properties of the tectorial membranes of wild-type and mutant mice using longitudinally propagating waves. Charlsie Lemons (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, charlsielem@gmail.com), Jonathan B. Sellon, Roberto D. Ortiz (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA), Elisa Boatti (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Dennis M. Freeman (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA), and Julien Meaud (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

The tectorial membrane (TM) is an important structural component of the mammalian inner ear. Examination of cochlear physiology in genetically modified mice has demonstrated that mutation of genes affecting TM proteins causes changes in key characteristics of cochlear function. Characterizing the differences in material properties between wild-type and mutant mice could provide insight into the source of reported differences in cochlear physiology. In this study, optical images of isolated TM segments of wild-type and genetically modified mice in response to harmonic radial excitation at acoustic frequencies are used to determine the amplitude and phase of the motion. Wave propagation on the TM segments is modeled using finite element models that take into account the anisotropy, viscoelasticity, and finite dimensions of the TM and the presence of a viscous boundary layer. An automated least-square fitting algorithm is used to find anisotropic, viscoelastic material parameters of wild-type and mutant mice TMs at acoustic frequencies. The resulting material properties are compared to previous estimates of the TM properties.

11:20

2aBA11. Lung ultrasound surface wave elastography for assessing interstitial lung disease. Xiaoming Zhang, Boran Zhou, Ryan Clay, Jinling Zhou, Thomas Osborn, Brian Bartholmai, James F. Greenleaf, and Sanjay Kalra (Mayo Clinic, 200 1st ST SW, Rochester, MN 55905, zhang.xiaoming@mayo.edu)

A lung ultrasound surface wave elastography (LUSWE) technique is developed for assessing interstitial lung disease (ILD). In LUSWE, a 0.1 second harmonic vibration is generated on the chest wall of a subject using a handheld vibrator. An ultrasound probe is aligned with the vibration indenter in the same intercostal region to measure the generated surface wave speed of the lung. A human subject is examined in a sitting position. The lung is tested at the total lung volume. The upper anterior lungs are tested through the second intercostal space in the mid-clavicular line. The lower lateral lungs are tested in the mid-axillary line and the lower posterior lungs are tested in the mid-scapular line. In a prospective clinical study, we measure both lungs through the six intercostal regions for patients and healthy controls. The surface wave speed is measured at 100 Hz, 150 Hz, and 200 Hz. Significant differences of wave speed between patients and controls were found in all lung regions at all frequencies. We also found positive correlation between LUSWE and CT analyses. LUSWE is a safe and noninvasive technique for generating and measuring surface wave propagation on the lung. LUSWE may be useful for assessing ILD.

11:35

2aBA12. Fingertip elastography. Altaf Khan (Mech. Eng., Univ. of Illinois at Chicago, Chicago, IL) and Thomas J. Royston (BioEng., Univ. of Illinois at Chicago, 851 South Morgan St., MC 063, Chicago, IL 60607, troyston@uic.edu)

Dynamic elastography methods attempt to quantitatively map soft tissue viscoelastic properties. Application to the fingertip, relevant to medical diagnostics and to improving tactile interfaces, is a novel and challenging application, given the small target size. In this feasibility study, an annular actuator placed on the surface of the fingertip and driven harmonically at multiple frequencies sequentially creates geometrically focused surface waves. These surface wave propagation patterns are measured using scanning laser Doppler vibrometry. Reconstruction (the inverse problem) is performed in order to estimate fingertip soft tissue viscoelastic properties. The study identifies limitations of an analytical approach versus an optimization approach that utilizes a finite element model. Measurement at multiple frequencies reveals limitations of an assumption of homogeneity of material properties. Identified shear viscoelastic properties increase significantly as frequency increases and the depth of penetration of the surface wave is reduced, indicating that the fingertip is significantly stiffer near its surface. Additional studies using Optical Coherence Tomography (OCT) and Magnetic Resonance Imaging (MRI) provide insights into wave propagation at near sub-surface and deep sub-surface zones.

TUESDAY MORNING, 8 MAY 2018 GREENWAY D, 9:00 A.M. TO 12:00 NOON

Session 2aEA

Engineering Acoustics and Physical Acoustics: Miniature Acoustic Transducers I

Vahid Naderyan, Chair
Physics/National Center for Physical Acoustics, University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677

Chair's Introduction—9:00

Invited Papers

9:05

2aEA1. A revisit to the lumped element model of capacitive MEMS microphones. Yu Du, Michael Kuntzman, and Yenhao Chen (Knowles Corp., 1151 Maplewood Dr., Itasca, IL 60143, Yu.Du@knowles.com)

Lumped element model (LEM) in the format of equivalent circuits are commonly used to describe dynamic characteristics of electromechanical transducers such as capacitive MEMS microphones. The derivation of such a LEM relies on classical electroacoustic and electromechanical principals and assumes physical elements with simple geometries and boundary conditions. However, practical designs rarely meet these assumptions, which leads to difficulties to extract model parameters with pure theoretical methods and limits the model usage to guide real designs. This study presents an improved LEM for MEMS microphones in which a hybrid theoretical-experimental method is used to extract model parameters. In particular, the mechanical compliance of the moving diaphragm and the transduction coefficient from the mechanical domain to the electrical domain are directly measured using laser Doppler vibrometer, which accounts for the influences of complex boundary conditions and geometrical shapes of actual designs. Furthermore, the diaphragm softening effect due to capacitive loading in the interface circuits is also included. Results predicted by the improved LEM are compared to measurements. It is shown that the newly proposed model can accurately capture the microphone response in the audio region. With care, its use may be extended to the ultrasonic range up to 40–50 kHz.

9:25

2aEA2. Multidomain modeling of acoustic transducers using SimScape. Thomas Blanford (Graduate Program in Acoust., The Penn State Univ., 1623 S. Ashwicken Ct, State College, PA 16801, teb217@psu.edu) and Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

Finite element modeling of acoustic transducers can capture many complex behaviors that are neglected in simpler analyses. However, the complexity that requires finite element modeling is often confined to a small region of the total device. In that case it is possible to combine the finite element model of the subsection that requires this complexity with dynamic system modeling methods for the balance of the design. Dynamic system modeling includes earlier methods of analog circuit modeling and generalizes those methods appropriately tor acoustic transducer analysis. This paper will describe SimScape modeling of transducer structures in a way that allows combined modeling with FEA where needed. Simple examples will be selected from among moving coil speakers, balanced armature transducers, condenser microphones, and piezoelectric devices.

9:45

2aEA3. Biologically inspired acoustic sensors: From insect ears to miniature microphones. James F. Windmill (Dept. of Electron. & Elec. Eng., Univ. of Strathclyde, 204 George St., Glasgow G1 1XW, United Kingdom, james.windmill@strath.ac.uk)

Taking inspiration from insect ears to develop an acoustic device is not new. Research to take inspiration from the highly directional, sub-wavelength, *Ormia ochracea* fly's ear to produce miniature directional microphones is well-known. Since the 1990s, researchers have tried to implement *Ormia* based micro-electro-mechanical systems devices, typically using standard silicon or similar microfabrication. Much of this time has been spent trying to circumvent the fact that *Ormia* evolved to hear one specific frequency, that of a calling cricket, and so is not a broadband system as you would require for an audio microphone. Further, building a silicon system inspired by the mechanics of insect ears leads to various compromises. This talk will discuss ongoing work at the University of Strathclyde on silicon microfabrication of acoustic devices inspired by the *Ormia* system. It will focus on efforts to create MEMS microphones designed with multiple resonance frequencies in the human vocal range. It will also describe ongoing research into the use of active feedback in MEMS acoustic systems, inspired by auditory hearing mechanisms. Finally, it will also look at the latest research by the Strathclyde team into the application of 3D microfabrication for the development of bio-inspired acoustic devices.

10:05

2aEA4. Miniature microphones: Non-standard fabrication approaches and associated modeling issues. Libor Rufer (TIMA Lab., Univ. Grenoble Alpes, France, 46, Av. Felix Viallet, Grenoble 38031, France, Libor.Rufer@univ-grenoble-alpes.fr)

Micro-electro-mechanical systems (MEMS) allowed miniature microphones that during the last few decades have left research laboratories to enter a massive industrial development. Their characteristics meet the most demanding specifications of various portable devices. This talk will show some non-standard approaches to the fabrication technology and associated modeling issues using two cases of specific designs in support. The first case is a piezoresistive microphone devoted to high-intensity and high-frequency acoustic fields. To damp the resonance, an atypical diaphragm structure was employed. Such a solution, allowing relatively large airgap required by large diaphragm displacements, will be presented and analyzed. The second case concerns a design of a microphone fabricated with a standard complementary metal oxide semiconductor (CMOS) technology. This technology, in contrast with dedicated technologies typically used by the microphone industry, is readily available to small companies and academic institutions. One of the drawbacks of the CMOS-MEMS approach is a relatively high degree of design constraints represented by materials and layers dimensions involved in the CMOS process. Moreover, the device layout must take into account specific requirements of the fabrication technique. This talk will present some of these constraints and their impact on the microphone performance that will be supported by simulation results.

10:25–10:40 Break

10:40

2aEA5. Spin-MEMS microphone based on highly sensitive spintronic strain-gauge sensors. Yoshihiko Fuji, Yoshihiro Higashi, Shiori Kaji (Corporate Res. & Development Ctr., Toshiba Corp., 1, Komukai-Toshiba-cho, Saiwai-ku, Kawasaki, Kanagawa 211-8583, Japan, yoshihiko.fuji@toshiba.co.jp), Kei Masunishi (Corporate Manufacturing Eng. Ctr., Toshiba Corp., Yokohama, Kanagawa, Japan), Tomohiko Nagata, Akiko Yuzawa, Kenji Otsu, Kazuaki Okamoto, Shotaro Baba, Tomio Ono, and Michiko Hara (Corporate Res. & Development Ctr. (Retired), Toshiba Corp., Kawasaki, Kanagawa, Japan)

We report a novel spintronic MEMS (Spin-MEMS) microphone, which is a new type of resistive microphone. For this microphone, spintronic strain-gauge sensors (Spin-SGSs) are integrated on a bulk micromachined diaphragm. The Spin-SGSs are based on magnetic tunnel junctions (MTJs) similar to those used as magnetic sensors in hard disk drives. In work to date, we have experimentally confirmed that the Spin-SGS exhibits a high gauge factor in excess of 5000, which is 100-fold that for a conventional poly-Si piezoresistor, by adopting a novel amorphous Fe-B-based sensing layer with high magnetostriction and low coercivity. Thanks to the high strain sensitivity of the Spin-SGSs, the Spin-MEMS microphone exhibits a signal-to-noise ratio (SNR) of 57 dB(A). A Spin-MEMS microphone with a first resonance frequency of over 70 kHz was also fabricated that exhibits an SNR of 49 dB(A), which is promising for acoustic health monitoring. In this study, we compared the operation sounds of defective and normal bearings using the Spin-MEMS microphone. The Spin-MEMS microphone detected differences in the operation sounds between the defective and normal bearings in the high-frequency range of 10 kHz to 50 kHz.

11:00

2aEA6. A MEMS condenser microphone based acoustic receiver for totally implantable cochlear implants. Flurin Pfiffner (Dept. of Otorhinolaryngology, Head and Neck Surgery, Univ. of Zurich, Univ. Hospital Zurich, Zurich, Zurich, Switzerland), Lukas Prochazka, Ivo Dobrev, Adrian Dalbert, Jae Hoon Sim (Dept. of Otorhinolaryngology, Head and Neck Surgery, Univ. of Zurich, Univ. Hospital Zurich, UniversitätsSpital Zürich, Frauenklinikstrasse 24, ORL Klinik, Zurich, Switzerland, Lukas.Prochazka@usz.ch), Francesca Harris (Cochlear Technol. Ctr., Mechelen, Belgium), Jeremie Guignard (Cochlear AG, Boston, Massachusetts), Joris Walraevens (Cochlear Technol. Ctr., Mechelen, Belgium), Christof Röösli, and Alex Huber (Dept. of Otorhinolaryngology, Head and Neck Surgery, Univ. of Zurich, Univ. Hospital Zurich, Zurich, Switzerland)

The goal of the present project is to develop intracochlear acoustic receivers (ICAR's) for measurement of the sound pressure in the inner ear of human temporal bones and in acute large animal experiments. In addition, the ICAR is designed to be used as an implantable microphone for totally implantable cochlear implant (TICI) systems. The presented ICAR concept is based on a commercially available MEMS condenser microphone customized with a protective diaphragm providing sealing properties and optimized sensor head geometry for accessing the tiny fluid-filled cavities of the human inner ear. The first ICAR prototypes (PT I) have been used for numerous intracochlear sound pressure measurements in human and sheep temporal bones. The data thus obtained are in good agreement with the literature. The second ICAR prototype (PT II) was further adapted for surgical insertion in the scala tympani in acute large animal experiments. First experiments have been successfully performed and further revealed that the presented ICAR concept is a suitable receiver technology for TICI systems. Currently, the development of a fully biocompatible ICAR (PT III) is ongoing. This sensor must fulfill all important requirements of a TICI device such as high performance, low power consumption and good system integration.

11:20

2aEA7. Survey of simulations techniques used in balanced armature transducers. Charles King (R&D, Knowles Electronics, 1151 Maplewood Dr, Itasca, IL 60302, charles.king@knowles.com)

Balanced armature transducers, also known as receivers, are commonly used in hearing aids, hearables, and in-ear monitors. These are difficult to simulate. They are very small devices, designed to fit inside the ear canal, and measurement of basic quantities is physically difficult. Additionally, multiple magnetic non-linearities are balanced to create a high-efficiency / high-output device. Spice simulations are well suited to understand the linear frequency response of the device. Finite element simulations are well suited to study steady state magnetic and geometric non-linearities. Time domain algebraic differential equation techniques can bridge these two simulation regimes. With this tool the amplitude response, frequency response, and distortion caused by the non-linearity's can be studied. The talk will examine strengths, weaknesses, and when it is appropriate to use each of these individual techniques.

11:40

2aEA8. Vibroacoustic simulation of hearing aid receivers. Brenno Varanda (Knowles Corp., 1410 Alexander Way, Clearwater, Florida 33756, bvarand1@gmail.com)

The overall aim of this research is to develop practical vibroacoustic models of hearing aid receivers, a key electro-acoustic component of hearing aids. The receiver is a high efficiency miniature sound source which utilizes a balanced armature electromagnetic motor. A standard side effect for most balance armature receivers is structural vibration. This receiver-borne structural vibration can travel through the hearing aid package to the microphones, resulting in undesirable oscillations, just like acoustic feedback. The receiver models are used to help hearing aid designers refine vibration isolation mounts and package components to reduce both acoustic and receiver-borne structural feedback. The model consists of a simplified electro acoustic circuit-equivalent that can easily be coupled to multi-physics finite element analyses. The model has been validated against standard hearing industry measurements and proved to be effective on predicting transmissibility forces across various speaker attachments.

Session 2aNS

Noise, Psychological and Physiological Acoustics, and Speech Communication: Hearing Health Across a Lifespan: Hearing Screening From Cradle to Grave

William J. Murphy, Cochair
Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998

Alexander L. Francis, Cochair
Purdue University, SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907

Chair's Introduction—8:30

Invited Papers

8:35

2aNS1. Hearing health across a lifespan: Hearing screening from cradle to grave. John Eichwald (National Ctr. for Environ. Health, Centers for Disease Control and Prevention (CDC), 1600 Clifton Rd. NE, MS-E18, Atlanta, GA 30333, jeichwald@cdc.gov)

The 2016 National Academies of Sciences, Engineering, and Medicine report "Hearing Health Care for Adults: Priorities for Improving Access and Affordability" included a call to action for government agencies to strengthen efforts to collect, analyze, and disseminate population-based data on hearing loss in adults. In partial response, the Centers for Disease Control and Prevention (CDC) analyzed the most recent available data collected both by questionnaire and audiometric tests of participants aged 20–69 years in the 2011–2012 National Health and Nutrition Examination Survey to determine the presence of audiometric notches indicative of noise-induced hearing loss. Prevalence of both unilateral and bilateral audiometric notches and their association with self-reported exposure to loud noise were calculated. Nearly one in four adults had audiometric notches, suggesting a high prevalence of noise-induced hearing loss. The prevalence of notches was higher among males. Almost one in four U.S. adults who reported excellent or good hearing had audiometric notches. Among participants who reported exposure to loud noise at work, almost one third had a notch. Noise-induced hearing loss is a significant, often unrecognized public health problem in the United States.

8:55

2aNS2. Hearing health across the lifespan: Prevention begins in early in life. Deanna K. Meinke (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, 501 20th St., Gunter Hall 1500, Campus Box 140, Greeley, CO 80639, Deanna.Meinke@unco.edu)

Hazardous noise exposures begin early in life and evidence of noise-induced hearing loss (NIHL) and tinnitus begins to appear in adolescence and early adulthood. In animal studies, these early noise exposures can cause permanent loss of greater than 50% of cochlear-nerve/hair-cell synapses that do not recover on their own and may contribute to greater age-related hearing loss in the future. All of the landmark studies over the past 75 years that identified the risk of NIHL in children and adolescents, ended with a recommendation for education or intervention to address the problem. In 1990, a consensus statement on Noise and Hearing Loss from the National Institutes for Health called for "a comprehensive program of education regarding the causes and prevention of NIHL should be developed and disseminated, with specific attention directed toward educating school-age children." Today, there continues to be a need to nationally coordinate public health efforts to systematically address the prevention of NIHL in youth using evidence-based hearing health promotion based upon health communication science. Dangerous Decibels® is a successful evidence-based intervention program that has been demonstrated to be effective in changing knowledge, attitudes, beliefs and behaviors of both youth and adults for the prevention of NIHL and tinnitus.

9:15

2aNS3. Hearing health for adults. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

Recent analyses of the National Health and Nutrition Examination Survey estimate that 14% of adults between ages 20 and 69 have hearing loss defined as average hearing thresholds worse than 25 dB HL across four frequencies 0.5, 1, 2, and 4 kHz. Noise exposure is a primary cause of preventable hearing loss in the United States; an estimated 22 million American workers are exposed to hazardous noise every year. Given its associated implications on communication, education, employment opportunities, job performance, injury-risk, depression, and anxiety—hearing loss places a significant burden on society. Preventive initiatives, workplace hearing conservation programs, and habilitation/rehabilitation efforts need to be tailored to specific populations, stages of life, and hearing risk factors. Innovative hearing health programs must be developed, evaluated, and implemented. This presentation will discuss innovations in hearing loss prevention and promote hearing health for adults in the occupational and non-occupational settings.

9:35

2aNS4. Development and implementation of a military auditory fitness-for-duty test battery. Jennifer B. Tufts (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT 06269, jennifer.tufts@uconn.edu), William Audette, Jed Wilbur, and Odile Clavier (Creare LLC, Hanover, NH)

Hearing loss is prevalent throughout the active duty military forces. However, there are no validated standards for assessing a service member's functional hearing ability and how it relates to his or her ability to accomplish the mission. In many cases, hearing-critical tasks take place in non-optimal environments with background noise or competing speech from multiple sources. Presently, hearing abilities are primarily assessed via pure-tone audiometry, without regard for the hearing-critical needs and noise environment of the individual. The development of a test battery to evaluate functional hearing ability is a highly sought-after goal. With support from the Department of Defense, Creare LLC and the University of Connecticut are developing the MILSINT, a new sound-recognition-in-noise test geared specifically for active duty personnel. The MILSINT is one component of an auditory fitness-for-duty test battery that at present also includes the Military HINT, a version of the Hearing In Noise Test (HINT) that uses military-specific phrases. The MILSINT and Military HINT are administered using a tablet interface and wireless sound-attenuating headset developed by Creare, LLC., making these tests highly portable and versatile. This presentation will describe the development of the test battery, its physical implementation, and performance data gathered to date.

9:55–10:10 Break

10:10

2aNS5. Progress in assessing and limiting occupational noise exposures in the United States. Richard Neitzel, Benjamin Roberts, Stephanie Sayler (Dept. of Environ. Health Sci., Univ. of Michigan, 1415 Washington Heights, 6611 SPH I, Ann Arbor, MI 48103, rneitzel@umich.edu), Wenting Cheng, and Bhramar Mukherjee (Dept. of Biostatistics, Univ. of Michigan, Ann Arbor, MI)

Occupational noise exposures in the United States (US) have resulted in a substantial occupational health burden on US workers. Despite the presence of regulations mandating exposure assessment and hearing loss prevention measures for highly-exposed workers, the prevalence of noise-induced hearing loss continues to be high. However, several recent efforts may result in progress in reducing exposures to, and health impacts from, occupational noise. First, the American Conference of Governmental Industrial Hygienists has recently proposed revisions to the organization's Threshold Limit Value (TLV) for Audible Sound, formerly referred to as noise. The proposed revised TLV includes updated documentation as well as notes related to non-auditory health effects of noise (including cardiovascular disease and injuries), which have not previously been considered in an occupational noise exposure limit. Second, a measurement-based national Job Exposure Matrix for Occupational Noise in the US and Canada has been developed at the University of Michigan and is now publicly available (noisejem.sph.umich.edu). This resource represents a useful tool for better understanding exposures to noise by job and industry. These two efforts, along with others, represent important progress in enhancing occupational health by protecting workers from the adverse health effects of noise.

10:30

2aNS6. Have the NIOSH age correction tables gone stale? Gregory Flamme (SASRAC, 2264 Heather Way, Forest Grove, OR 97116, gflamme@sasrac.com), Kristy Deiters (SASRAC, Portage, MI), William J. Murphy, Christa Themann (National Inst. for Occupational Safety and Health, Cincinnati, OH), and Mark R. Stephenson (SASRAC, Loveland, OH)

Occupational Hearing Conservation programs permit subtracting cross-sectional trends in hearing sensitivity from the changes observed with an exposed worker. Regulatory agencies in the U.S. define the expected cross-sectional trend using the NIOSH age correction tables, which summarized mean differences in comparatively small groups of men and women 50 years ago. At all ages, hearing sensitivity is better now than it was 50 years ago, and important demographic characteristics (e.g., race/ethnicity) that predict cross-sectional trends were not included in the NIOSH tables. Quantile regression results from the National Health and Nutrition Examination Survey (NHANES) were used to derive revised age correction tables that can be applied to men or women across a wider range of ages and race/ethnicity categories. These age correction tables and comparisons with the prior NIOSH tables will be presented.

10:50

2aNS7. Individual differences in suprathreshold hearing and relationship to cochlear synaptopathy. Hari M. Bharadwaj (Speech, Lang., & Hearing Sci., and Biomedical Eng., Purdue Univ., 715 Clinic Dr., Lyles-Porter Hall, West Lafayette, IN 47907, hbharadwaj@purdue.edu)

Threshold audiometry, although the foundation of current clinical hearing evaluations, provides limited information about suprathreshold hearing. Over the last few years, we have documented that even among listeners with normal thresholds and no hearing complaints, large individual differences exist in the ability to perceive subtle temporal features of clearly audible sounds, and to selectively process target speech in the presence of competing sounds. Furthermore, we find that these suprathreshold perceptual differences correlate with suprathreshold physiological measures from the brainstem and auditory nerve suggesting that perceptually-relevant differences may be present early along the auditory pathway. As a candidate mechanism explaining these observations, animal studies of acoustic overexposure and aging have robustly demonstrated that the afferent synapses and nerve terminals innervating the cochlea are especially vulnerable to damage. Interestingly, even a significant loss of cochlear synapses ("synaptopathy") does not lead to changes in audiometric thresholds. However, synaptopathy, if present in humans may contribute to differences in suprathreshold hearing especially in noisy environments. This presentation will summarize human evidence that is indirectly in support of the notion, and describe our ongoing efforts to test this hypothesis further and translate markers for synaptopathy from small laboratory animals to humans.

Contributed Papers

11:10

2aNS8. The non-auditory health effects of chronic noise exposure: A review. Jennifer Scinto (SoundSense, LLC, PO Box 1360, Wainscott, NY 11975, jennifer@soundsense.com), Bonnie Schnitta (SoundSense, LLC, East Hampton, NY), John Durant, and Neelakshi Hudda (Civil and Environ. Eng., Tufts Univ., Medford, MA)

This paper serves as a review of the most recent literature on the effects of chronic noise exposure on the health of human listeners. The auditory effects of occupational and other exposure to noise have been well researched and the prevention and treatment of noise-induced hearing loss is largely understood. However, as the populations of urban areas and large city centers grow, the interest in the non-auditory health effects of exposure to environmental noise such as transportation noise, aircraft noise, and other community noise has also increased. Within the past few years, several new studies have been conducted strengthening the evidence for a link between exposure to high levels of environmental noise and ill-health, especially with regards to cardio-vascular and endocrine health, immune function, sleep loss, and mental health. The research covered in this review is inclusive of human experimental, epidemiological, and mechanistic studies. Additional research will be needed to completely assess and understand the increasingly present threat to human health that is chronic environmental noise. This review will be used as the foundation for a future study examining the correlation between ill-health and noise pollution from a nearby airport.

11:25

2aNS9. Psychophysiological responses during cognitively demanding work in subjectively annoying background noise. Jordan N. Oliver, Weonchan Sung, Patricia Davies, and Alexander L. Francis (Purdue Univ., Speech, Lang. & Hearing Sci., Purdue University, West Lafayette, IN 47907, oliver49@purdue.edu)

People who work in noisy environments are at greater risk for stress-related diseases, including hypertension and stroke, even when noise levels are too low to damage hearing. Such noise may be harmful, especially to noise-sensitive individuals, because the psychological annoyance that it causes induces physiological stress responses that are damaging to health over the long term. This study was designed to investigate the link between subjective noise annoyance and physiological measures of arousal and displeasure due to the presence of background noise. Cardiovascular, electrodermal, respiratory, and facial muscular activity were recorded from 32 listeners during the completion of a demanding working memory task under different listening conditions. Participants completed four levels of memory task demand in silence and in two different continuous noises similar to that produced by HVAC equipment. Both noises were comparable in terms of loudness and presentation level (54–60 dBA) but differed in perceived annoyance (based on ratings from a panel of listeners in a previous study) and in acoustic properties associated with noise annoyance (roughness, tonality, and sharpness). Behavioral measures of memory task performance, noise sensitivity, personality traits, and subjective effort will be presented and related to physiological measures, and implications for future research will be discussed.

TUESDAY MORNING, 8 MAY 2018 GREENWAY J, 9:40 A.M. TO 11:25 A.M.

Session 2aPA

Physical Acoustics: Infrasound for Global Security I

Philip Blom, Chair
Los Alamos National Laboratory, Los Alamos National Laboratory, PO Box 1663, Los Alamos, NM 87545

Chair's Introduction—9:40

Invited Papers

9:45

2aPA1. Reflections on Los Alamos infrasound research. Rodney W. Whitaker (Earth and Environ. Sci., Los Alamos National Lab., EES 17 MS F665, LANL, Los Alamos, NM 87545, rww@lanl.gov)

Researchers at Los Alamos National Laboratory began work in infrasound research in 1982 under Department of Energy funding. This beginning was to investigate the generation and propagation of surface ground motion generated atmospheric infrasound signals. Infrasound arrays were operated to measure the infrasound signals at near regional distances. The program was successful and continued until the end of US underground nuclear testing in 1992. During this time, data were collected on earthquakes, bolides and on various man-made high explosive tests. As work on the Comprehensive Nuclear Test Ban Treaty was continuing, Los Alamos, working with Sandia National Laboratory, fielded a prototype infrasound array following Comprehensive Nuclear Test Ban Treaty guidelines, which operated from mid 1997 to 2007. More recent work emphasizes seismo-acoustic research in explosion monitoring, improving infrasound propagation models for better source location and analysis and the development of Python based tools. In this talk, I will highlight some aspects of the earlier and current Los Alamos activity.

10:05

2aPA2. Progresses with the International Data Centre Infrasound System. Pierrick Mialle (CTBTO, Vienna Int. Ctr., P.O. Box 1200, Vienna 1400, Austria, pierrick.mialle@ctbto.org) and Nimar Arora (Bayesian Logic, Inc., Union City, CA)

The International Data Centre (IDC) advances its methods and continuously improves its automatic system for infrasound technology. The IDC focuses on enhancing the automatic system for the identification of valid signals and the optimization of the network detection threshold by identifying ways to refine signal characterization methodology and association criteria. An objective of this study is to reduce the number of associated infrasound arrivals that are rejected from the automatic bulletins when generating the reviewed event bulletins (REB). Progresses related to several ongoing projects at the IDC will be review:—improving the detection accuracy at the station processing stage by introducing the infrasound signal detection and interactive review software DTK-(G)PMCC (Progressive Multi-Channel Correlation) and by evaluating the performances of detection software;—development of the new generation of automatic waveform network processing software NET-VISA to pursue a lower ratio of false alarms over GA (Global Association) and a path for revisiting the historical Infrasound Reference Event Database (IRED). The IDC identified a number of areas for future improvement of its infrasound system that will be addressed here.

10:25

2aPA3. What does infrasound signal duration tell us about signal propagation distance and waveguide structure? David Green and Alexandra Nippress (AWE Blacknest, Brimpton Common, Reading RG7 4RS, United Kingdom, dgreen@blacknest.gov.uk)

A global network of microbarograph arrays is currently being constructed to support verification of the Comprehensive Nuclear-Test-Ban Treaty. The identification of explosively generated infrasound signals that have propagated thousands of kilometers to the arrays, and the subsequent association of signals across the sparse network, remains a challenge. One signal parameter that has not been extensively studied, but may assist in source identification procedures, is the signal duration. The durations of 42 high signal-to-noise ratio signals from 35 near-surface explosions, recorded at distances of between 25 and 6300 km, exhibit a weak relationship with source-to-receiver range; longer propagation paths result in longer signal durations. The variation in signal duration at a given source-to-receiver range depends upon the atmospheric waveguide structure. At propagation distances greater than 2000 km, long duration signals are generated within weak waveguides characterized by a small excess in stratospheric effective sound speed compared to that at the ground surface. These waveguides permit signal propagation with high celerity (up to 335 m/s). Shorter duration signals at these distances occur in stronger waveguides that exhibit greater along-path variability; these signals do not exhibit celerities greater than 315 m/s. The utility of signal duration within the context of explosion monitoring is explored.

10:45

2aPA4. Exploiting both output channels of seismically decoupled infrasound sensors for enhanced transient detection and direction-of-arrival estimation. William G. Frazier (Hyperion Technol. Group, Inc., 3248 West Jackson St., Tupelo, MS 38801, gfrazier@hyperiontg.com)

Some infrasound sensors have internal sensing capabilities that remove or permit subsequent removal of much of the effect of ground motion on the measured pressure fluctuations. This presentation explores possible enhancements to transient acoustic signal detection and direction-of-arrival estimation when data corresponding to the ground motion and pressure fluctuation signals are analyzed simultaneously using statistical array signal processing techniques. Data from three 4-element infrasound sensor arrays that were deployed to detect explosive events occurring from 10 km to 20 km distant are used as the basis for the study.

11:05

2aPA5. The application of acoustic particle velocity sensors to airborne infrasonic signals. W. C. Kirkpatrick Alberts (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20723, william.c.alberts4.civ@mail.mil)

Infrasound arrays, because of the wavelengths involved and the necessity for minimal spatial aliasing, can be quite large, e.g. an array layout often used for beamforming for frequencies up to 10 Hz is a centered, 17-m radius equilateral triangle. Available real estate is often limited such that optimally sized arrays are not possible. An acoustic particle velocity sensor, in principle, can perform the same job as a large infrasound array, but in a much smaller package (less than the size of a single infrasound sensor) since it provides, in post processing, a direct measurement of the wavefront normal. However, localized meteorological perturbations to the wavefront normal could cause significant bearing estimate errors when using a particle velocity sensor. Recently, as part of the characterization of a mobile infrasound source and during the collection of sounds from space station resupply rockets, acoustic particle velocity sensors were positioned next to infrasound sensors in order to test the concept of using these sensors for accurate pointing at infrasound frequencies. Results from these experiments will be discussed.

Session 2aPP

Psychological and Physiological Acoustics, Speech Communication, and Signal Processing in Acoustics: Phase Locking and Rate Limits in Electric Hearing

Mathias Dietz, Chair
National Centre for Audiology, Western University, 1201 Western Road, London, ON N6G 1H1, Canada

Chair's Introduction—8:00

Contributed Paper

8:05

2aPP1. The effects of sound coder carrier rate and modulation bandwidth on voice pitch perception in cochlear implant users. Damir Kovacic (Depertment of Phys., Univ. of Split, Faculty of Sci., R.Boskovica 33, Split 21000, Croatia, Damir.Kovacic@pmfst.hr) and Chris James (Cochlear SAS France, Toulouse, France)

We employed the dual filter-bank "STEP" coder to separately control the spectral and temporal modulation resolution of analysis channels. Previously we compared vowel pitch ranking and gender classification with eight subjects using enhanced modulation at F0—including across-channel synchronised modulation to the ACE coder. There was no significant improvement using modulation enhanced coding versus ACE across subjects. In a follow-up experiment we looked at the effect of stimulation rate on voice pitch perception. Since there are large inter-subject differences in overall temporal pitch acuity we hypothesised that some subjects' performance may be more greatly influenced by carrier rate than others, or that some subjects may find sound quality satisfactory with lower carrier rates than those in their clinical processors. We used a version of STEP with a very short temporal envelope analysis window of 2 ms which allows a very low latency real-time processing implementation and large maximum modulation bandwidth. Subjects were tested using carrier rates of 1000, 500 and 250 pps/ch with modulation bandwidths controlled via low-pass filtering. Pilot data indicated that the new low-latency coder provides very good sound quality compared to ACE using 1000 pps/ch or 500 pps/ch. Also the modulation bandwidth could be tuned at different carrier rates to optimize voice pitch perception based on temporal cues. This opens the potential for lower stimulation rates to be used in CI coding while maintaining optimal temporal resolution.

Invited Papers

8:20

2aPP2. Phase-locking in acoustic hearing: Upper limits, enhancement, and transformations. Philip X. Joris (NeuroSci., Univ. of Leuven, Lab Auditory Neurophysiol, Herestraat 49 box 1021, Leuven B-3000, Belgium, philip.joris@med.kuleuven.be)

Phase-locking is a remarkable neural property of various mechanoreceptor-based sensory systems, and the neural coding of cochlear vibrations is the primary model system to study this phenomenon. Transduction and synaptic transmission by cochlear inner hair cells enables but also limits temporal coding in the auditory nerve. I will review phase-locking to sound fine-structure in the auditory nerve of various species, including humans, assessed with different types of sound stimuli and using different metrics. The upper-frequency limit of phase-locking in cell populations in the central nervous system is lower than that in the auditory nerve, and at the same time, the nature of the temporal code is transformed in these neurons: phase-locking is enhanced in that it is more consistent and has decreased jitter, and it has profound effects on the average discharge rate of various populations of neurons, both monaurally and binaurally. Similar phenomena (changing upper limits, enhancement, and rate effects) are observed in the temporal coding of envelopes and click trains. While it is increasingly clear that many cell groups in the auditory brainstem have specialized mechanisms towards temporal coding, the relationship of the resulting response properties to perception remains unclear.

8:40

2aPP3. Comparison of temporal processing in the auditory brainstem neurons between acoustic and electrical hearing. Michaela Müller, Barbara Beiderbeck, Benedikt Grothe, and Michael Pecka (Biology II; Neurobiology, Ludwig-Maximilians Univ. Munich, Grosshaderner Strasse 2, Martinsried 82152, Germany, pecka@bio.lmu.de)

Spatial hearing is essential for communication as well as navigation in everyday life. Unfortunately, sound localization in these complex environments is severely limited in patients with bilateral cochlear implants (CIs) and its restoration thus remains one of the central obstacles of CI research. Spatial sensitivity is generated by neurons in the brainstem that detect differences in the arrival time between excitatory and inhibitory inputs from the two ears on the scale of only microseconds. However, the mechanisms underlying this precise temporal integration of individual inputs and potential differences in temporal precision during electrical stimulation are still not understood. Here, we obtained *in vivo* electrophysiological recordings from single neurons in the brainstem of Mongolian Gerbils

(*Merionesunguiculatus*). We characterized the temporal precision of action potential firing in response to click trains in binaural neurons of the Lateral Superior Olive and their upstream inputs (auditory nerve, cochlear nucleus and medial nucleus of the trapezoid body). To assess potential differences in temporal precision between acoustic and electrical stimulation, complementary data are obtained in normal hearing animals and animals that underwent cochlear implantation. Ultimately, we aim to identify basic principles of binaural integration in the auditory brainstem during acoustic and CI-based spatial hearing.

9:00

2aPP4. Measurements and models of auditory nerve spike rate and timing to electrical stimulation. Ian C. Bruce (Dept. of Elec. & Comput. Eng., McMaster Univ., Rm. ITB-A213, 1280 Main St. W, Hamilton, ON L8S 4K1, Canada, ibruce@ieee.org)

Physiological measurements of the response of auditory nerve (AN) fibers to electrical stimulation from a cochlear implant (CI) have in general exhibited much higher maximal entrainment rates and higher temporal precision than is observed for acoustical stimulation. However, it appears that this superior rate coding in the AN for electrical stimulation does not translate to better coding of rate in the midbrain or for the percept of temporal pitch. Similarly, the enhanced temporal precision for electrical stimulation does not lead to improved coding of interaural timing differences. In this talk, I will review physiological data and computational model simulations that indicate that the superior spike rate and timing representation for CI stimulation may only hold when considering the response of *some* single AN fibers to *repeated identical stimuli*. In fact, there is a large heterogeneity in spiking responses that can change dramatically as a function of stimulus current level. A multi-compartmental computational model will be used to demonstrate how the electrode- fiber geometry and the membrane biophysics could be contributing to these phenomena. I will also discuss the implications of the likely population response of the AN to CI stimulation for brainstem processing. [Work supported by NSERC Discovery Grant 261736.]

9:20

2aPP5. Effect of site of spike generation on temporal coding in the electrically stimulated auditory nerve. Suyash N. Joshi (Hearing Systems Group, Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Bygning 352, Kongens Lyngby 2800, Denmark, sjoshi@elektro.dtu.dk)

At least two sites of spike generation are observed in the electrically stimulated auditory nerve, namely, the peripheral axon and the central axon. The peripheral axon responds to the cathodic charge and the central axon to the anodic charge with an approximate spike-latency difference of 200 μs. The peripheral axon also shows longer refractory periods than the central axon. In this study, a phenomenological model of the electrically stimulated auditory nerve described in Joshi *et al.* [JARO 18(2), pp. 323–342] was used to understand the effect of spike generation sites on the temporal coding. Modulation detection thresholds for different carrier pulse rates were predicted from simulated responses of the auditory nerve, and refractoriness of both the axons was varied to test its effect on temporal coding. The results show that an increase in pulse rate increased the number of spikes generated at the central axon, hence decreasing the modulation detection thresholds. Refractoriness also played a crucial role in encoding the stimulus envelope such that extended refractory periods resulted in enhanced envelope coding. Increase in refractoriness at the central axon improved the envelope encoding. Implications of these findings for developing new stimulation strategies will be discussed.

9:40–9:55 Break

9:55

2aPP6. Modeling Cochlear Implant processing and bilateral performance in acoustic environments with multiple sources. H. Steven Colburn (Hearing Res. Ctr.; Biomed. Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, colburn@bu.edu)

The work reported here focuses on the use of cochlear implants in situations with multiple sources, especially speech sources, and in other difficult environments. After a review of previous measurements and models of cochlear implant (CI) stimulation, recent modeling studies of speech processing in conditions with multiple inputs in complex acoustic environments will be discussed. As noted above, complex environments considered include conditions with multiple speech sources, and this talk focuses on effects of spatial separation and voice pitch differences in confusions related to source separation. These effects include both energetic and informational masking. Multiple implant factors will be addressed, including interaural variability in the timing of the left and right stimuli, the effects of independent left and right automatic gain control, and the interactions with and basic role of pulse rate in the implants. [Work supported by NIDCD 5 R01 DC000100.]

10:15

2aPP7. Comparisons of electric and acoustic ITD coding in normal hearing gerbils. Maike Vollmer (Otolaryngol., Otto von Guericke Univ. Med. School, Leipziger Strasse 44, Magdeburg 39120, Germany, maike.vollmer@med.ovgu.de)

Small differences in the arrival time of sound between the two ears [interaural time differences (ITDs)] provide important cues for directional hearing and speech understanding in noise. Deaf subjects with bilateral cochlear implants (CIs) show relatively poor sensitivity to ITDs. It is unclear whether these limitations are due to differences in binaural brain circuits activated by electric compared to acoustic stimulation, mismatches in electric activation site across ears, or deafness-induced degradations in neural ITD processing. To identify potential limitations of electric ITD coding, we compared electric and acoustic ITD coding (i.e., ITD tuning and discrimination thresholds) in the same population of auditory brainstem and midbrain neurons in normal hearing gerbils. When compared in the same neurons, ITD coding to acoustic stimulation did not predict coding to electric stimulation. However, on a population level, neurons demonstrated surprising similarities in acoustic and electric ITD processing. Importantly, even short periods of deafness (2 weeks) severely degraded electric ITD processing. The findings suggest that discrepancies in ITD discrimination between bilateral CI users and normal hearing listeners are primarily due to deafness-induced changes in neural ITD processing rather than differences in the binaural brain circuits activated by either electric or acoustic stimulation.

10:35

2aPP8. Towards increasing timing sensitivity in electric hearing. Bernhard Laback (Acoust. Res. Institute, Austrian Acad. of Sci., Wohllebengasse 12-14, Vienna 1040, Austria, bernhard.laback@oeaw.ac.at), Sridhar Srinivasan (Acoust. Res. Institute, Austrian Acad. of Sci., Wien, Wien, Austria), Martin Lindenbeck, Maike Ferber (Acoust. Res. Institute, Austrian Acad. of Sci., Vienna, Austria), and Piotr Majdak (Acoust. Res. Instiute, Austrian Acad. of Sci., Wien, Austria)

Sensitivity of bilateral cochlear-implant (CI) listeners to interaural time differences (ITDs) in electric pulse trains is degraded compared to normal-hearing (NH) listeners presented with ITDs in pure tones. This degradation manifests both as an elevated ITD threshold and upper perceptual limit of stimulation rates. Similar limitations were observed for temporal pitch, despite the difficulty to disentangle temporal and place pitch cues in NH listeners. We tested the hypothesis that ITD and monaural rate-pitch sensitivity of CI listeners at high rates of electric stimulation can be improved by introducing extra pulses with short inter-pulse intervals (SIPIs) at low rates in amplitude modulated high-rate pulse trains. Results show that SIPIs significantly improved ITD and monaural rate-pitch sensitivity at low modulation depths. These similar improvements suggest a possible overlapping benefit of SIPIs for CI listeners in everyday environments requiring high rates for encoding speech. Considering the documented effects of neural deprivation, its potential reversal by training, and the potential for NH-like timing sensitivity as observed in exceptional CI listeners, we began to investigate the effects of vision-induced ITD training in CI listeners. Preliminary results will be discussed in the light of the timing sensitivity limitations in electric hearing.

10:55

2aPP9. Restoring binaural and spatial hearing in cochlear implant users. Ruth Litovsky, Alan Kan, Tanvi Thakkar, Sean R. Anderson (Waisman Ctr., Univ. of Wisconsin - Madison, 1500 Highland Ave., Waisman Ctr. Rm. 521, Madison, WI 53705, litovsky@waisman.wisc.edu), Z. Ellen Peng (Waisman Ctr., Univ. of Wisconsin - Madison, Wisconsin, U.S. Minor Outlying Islands), and Thibaud Leclère (Waisman Ctr., Univ. of Wisconsin - Madison, Madison, WI)

Adults and children who receive bilateral cochlear implants (BiCIs) have the potential to benefit from the integration of inputs arriving at the brain from both ears. Several factors play a key role in determining if patients will demonstrate binaural sensitivity. We are exploring these factors using two experimental approaches. In the first approach, BiCI users receive pulsatile stimulation to specific pairs of electrodes using research processors that synchronize stimulation with fidelity. We vary stimulus parameters such as temporal fine structure and envelope cues, places of stimulation along the cochleae, and number of electrodes to find parameters that maximize sensitivity to interaural differences for each patient. We also investigate the role of the electrode-neuron interface which is affected by numerous factors including neural health. In a second stimulation approach, we use clinical speech processor to deliver binaural stimulation designed specifically for that patient based on their clinical MAP. In these studies, we are using both standard psychophysics and eye gaze paradigms to understand the underlying processing involved in binaural and spatial hearing. This combined approach is enabling us to design multi-channel multi-rate stimulation strategies aimed at restoring binaural sensitivity and preserving speech understanding.

11:15

2aPP10. Temporal constraints on neural and psychophysical sensitivity to interaural level difference cues to sound source location. Andrew D. Brown and Daniel J. Tollin (Dept. of Physiol. & Biophys., Univ. of Colorado School of Medicine, University of Colorado Anschutz Medical Campus, 12631 E 17th Ave., Aurora, CO 80045, Daniel.Tollin@ucdenver.edu)

Interaural differences in the timing (ITD) and level (ILD) of impinging sounds carry critical information about source location. However, in everyday listening environments, sounds are often decorrelated between the ears by reverberation and background noise, degrading the fidelity of both ITD and ILD cues. Similar distortions to ITD and ILD are also experienced by hearing-impaired humans who use hearing aids or cochlear implants, as these devices also degrade temporal and intensive features of the signal at each ear. Here, we demonstrate that behavioral ILD sensitivity (in humans) and neural ILD sensitivity (in single neurons of the chinchilla auditory midbrain) remain robust under stimulus conditions that render ITD cues undetectable. Neural and behavioral data were compared to the outputs of a model of ILD processing with a single free parameter, the duration of excitatory-inhibitory interaction. Behavioral, neural, and modeling data collectively suggest that ILD sensitivity depends on binaural integration of excitation and inhibition within a $\gtrsim$3-ms temporal window, significantly longer than observed in lower brainstem neurons. This relatively slow integration potentiates a unique role for the ILD system in spatial hearing that may be of particular importance when informative ITD cues are unavailable. [Work supported by F32-DC013927 [ADB] and R01-DC011555 [DJT].]

11:35–11:55 Panel Discussion

Session 2aSC

Speech Communication, Animal Bioacoustics, and Psychological and Physiological Acoustics: Adapting Methods and Models for Vocal Production Across Human and Non-Human Primate Species

Benjamin Munson, Cochair
University of Minnesota, 115 Shevlin Hall, Minneapolis, MN

Michael B. Wilson, Cochair
Acoustics, Penn State, 1649 Highlandon Ct, State College, PA 16801

Mary E. Beckman, Cochair
Department of Linguistics, Ohio State University, 1712 Neil Ave., Columbus, OH 43210

Chair's Introduction—8:40

Invited Papers

8:45

2aSC1. Tracking chimpanzee pant-hoot changes across time and space. Michael Wilson (Anthropology, Univ. of Minnesota, 395 Humphrey Ctr., 301 19th Ave. S, Minneapolis, MN 55455, wilso198@umn.edu), Lisa R. O'Bryan (Dept. of Ecology, Evolution and Behavior, Univ. of Minneapolis, Sugar Land, Texas), Andrew R. Plummer (Comput. Sci. and Eng., Ohio State Univ., Columbus, OH), Mary E. Beckman (Linguist, Ohio State Univ., Columbus, OH), and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minneapolis, Minneapolis, MN)

Humans differ strikingly from other primates in the capacity for vocal learning. How and why such vocal flexibility evolved remains puzzling. Evidence of geographic variation in the pant-hoot calls of chimpanzees (*Pan troglodytes*) suggests that chimpanzees have some capacity for vocal learning, which is intriguing given the close phylogenetic relationship between humans and chimpanzees. Many questions remain, however, about the various factors that might contribute to variation in acoustic structure of pant-hoot calls, including body size, health, genetic relatedness, and within-individual variation. We are currently examining these factors in a study of longitudinal recordings from individuals in two neighboring chimpanzee communities in Gombe National Park, Tanzania. As in studies of sound change in human social groups, we need to understand the articulatory mechanisms that produce the vocalizations, to interpret acoustic variation against the backdrop of factors including body size, sex, and age. The same considerations in studies aimed at understanding the ontogeny of human vocalizations have prompted the development of age-specific articulatory synthesis models. We are adapting such models to design analysis-by-synthesis methods for our cross-population longitudinal study, and have organized this session to explore analogous methods for studying other vocalization types in chimpanzees and other non-human primate species.

9:05

2aSC2. Supervised and unsupervised machine learning approaches to classifying chimpanzee vocalizations. Nisarg P. Desai (Anthropology, Univ. of Minnesota, 395 Humphrey Ctr., 301 19th Ave. S, Minneapolis, MN 55455, desai054@umn.edu), Clarence Lehman (Ecology, Evolution, and Behavior, Univ. of Minnesota, Saint Paul, MN), Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), and Michael Wilson (Anthropology, Univ. of Minnesota, Minneapolis, MN)

Quantitative tools for classifying vocal repertoires have been constantly evolving with developments in machine learning and speech recognition research as well as increasing computing power. There are two main methodological considerations in classifying vocalizations: (i) choosing the classification technique and (ii) choosing the features for classification. Current state-of-the-art classification techniques are artificial neural networks (ANNs), support vector machines (SVMs), and ensemble methods like random forests (RFs). Current state-of-the-art features from speech recognition research include mel frequency cepstral coefficients (MFCCs). Bioacoustics researchers have applied these tools to problems including individual-, species-, and call-type identification, and vocal repertoire classification. However, researchers studying non-human primate vocalizations have only recently started adopting these approaches and none have applied them to study chimpanzee vocalizations. Here, we analyze vocalizations recorded in Gombe National Park, Tanzania. First, we use supervised classification techniques (ANNs, SVMs, and RFs) that involve training the models based on predefined call-types to evaluate the classification accuracy. Second, we use unsupervised techniques (that do not require prior knowledge of call-types), namely, K-means clustering, and self-organizing neural networks to identify discrete call types. We discuss the results from both supervised and unsupervised techniques and their strengths over traditional methods.

9:25

2aSC3. Acoustic communication by vocal tract modulation. Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

In both human and nonhuman animals, the airway system may serve as an instrument for acoustic communication. Flow-induced tissue vibrations and noise sources generate the acoustic excitation, whereas the configuration of the vocal tract provides a variable resonant filter system that transforms the excitation into a "message." This presentation will describe the development of a vocal tract model in which the effects of articulatory movements that produce speech are generated by specifying independent acoustic events along a time axis. These events consist of directional changes in the first three resonance frequencies of an acoustically-neutral airway configuration and are transformed, via acoustic sensitivity functions, into time-varying modulations of the vocal tract shape. The duration of each event may be considerably overlapped in time with other events to produce efficient transmission of information through the effects of coarticulation. The model will be used to demonstrate construction of syllables, words, and phrases based on a range of underlying idiosyncratic "neutral'' vocal tract configurations representative of typical and unusual airway systems. [Research supported by NIH R01-DC011275, NSF BCS-1145011, and NIH R01-DC006282.]

9:45

2aSC4. Articulatory modeling of human and non-human vocal production. Kiyoshi Honda and Ju Zhang (School of Comput. Sci. and Technol., Tianjin Univ., 135, Yaguan Rd., Jinnan Dist., Tianjin 300350, China, khonda@sannet.ne.jp)

2a TUE. AM

This paper reviews two of the many questions that might be addressed by adapting articulatory models of vocal production across primate species. The first concerns vocal-fold shaping at phonation. The human folds have been modeled to have uniform cross-sectional shapes with round edges along the longitudinal axis, whereas MRI slices reveal an obvious vertical prominence of the mucosa near each vocal process. This shape, due to depressed vocal processes in adduction, somewhat resembles the pointed edges of the vocal membranes in nonhuman primates, which are incorporated in some models of chaotic phonation in those species. This suggests a model of vocal-fold vibration propagating back to the posterior glottis, or coupled oscillation for irregularities in human voices. The second concerns tongue-lip coordination. In human vowel inventories, tongue retraction typically accompanies rounding. In a case report, a trained Chimpanzee was able to produce back vowels /a, o, u/ in whispering, but could not coordinate these articulatory positions with voiced phonation. This suggests an innate neural circuit for patterned tongue-lip coordination as seen in sucking, and such a basic neural unit may be active in humans to simplify the articulatory control of back rounded vowels. [Work supported by NSFC No. 61573254.]

10:05 – 10:20

10:20

2aSC5. The challenges of developing articulatory synthesis models of early vocal production in humans. Andrew R. Plummer (Ohio State Univ., 2015 Neil Ave., Columbus, OH 43210, plummer.321@osu.edu)

At birth, human infants can cry, but they cannot articulate anything like the sound patterns of their mothers' speech that they have been hearing in the last month or so in the womb. Learning to speak involves a re-tuning of the muscle systems for breathing, crying, and suckling in order to be able to articulate a rapid sequence of exquisitely coordinated movements of the speech articulators—gestures of the larynx, velum, tongue, and lips—in synchrony with a lengthened expiratory phase of the breathing cycle, to acoustically shape the air flowing from the lungs to produce the sound patterns that are the words of a specific language. Models of speech production developed using data on articulator movement and coordination in adults cannot be applied directly, because this learning takes place in the context of substantial changes to the size and shape of the vocal tract over the course of physical maturation. This talk describes the changes in vocal tract morphology in human development, and then reviews how articulatory synthesis systems developed for adults have been adapted to model the expanding repertoire of vocalizations that have been observed in infants with normal hearing over the first year of life.

10:40

2aSC6. Physical growth drives vocal sequence development in marmoset infants. Yisi Zhang and Asif Ghazanfar (Princeton Univ., Princeton Neurosci. Inst., Princeton, NJ 08544, yz9@princeton.edu)

The vocal sequences of marmosets exhibit many drastic changes during infancy. One of the most pronounced of these is the gradual convergence from a variety of call types in the undirected context to long distance contact calls (aka phee calls). We conjecture that such change is attributed to the interplay between the neural fluctuation and the biomechanics of the developing vocal periphery. To explore such possibility, we first quantitatively characterize vocal sequences and correlate them with animal momentary arousal levels. Based on this, we show that the diversity of respiratory and laryngeal patterns, as well as the commonly produced sequence transitions, can be generated via simple coupled oscillators with a low-dimensional input. In addition, the model predicts that as the respiratory apparatus grows, not only does the oscillatory period increase, but also the proportions of call types become biased towards phee calls. We further empirically test this hypothesis by placing the marmoset infants in a helium-oxygen (heliox) environment where the much lighter air reduces vocal effort, simulating smaller lungs. Consistent with the model prediction, the heliox manipulation reduces phee call duration and increases the proportion of other call types.

11:00

2aSC7. Chimpanzee food-associated calls display low referential potential in a wild population. Lisa R. O'Bryan (Ecology, Evolution and Behavior, Univ. of Minnesota, 13915 Lynnwood Ln, Sugar Land, TX 77498, obry0017@gmail.com), Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), and Michael Wilson (Ecology, Evolution and Behavior, Univ. of Minnesota, Minneapolis, MN)

Studies of non-human primate vocal communication commonly focus on "functionally referential signals," which are thought to function like human words, informing receivers about stimuli in the external environment. Captive studies of the food-associated "rough-grunt" of chimpanzees (*Pan troglodytes*) report that their acoustic structure varies according to food quality and even type, suggesting the existence of functionally referential communication in humans' evolutionary cousins. Nevertheless, studies of wild chimpanzees have produced mixed evidence that rough-grunts function referentially in natural contexts. The current study builds upon these findings by conducting an acoustic analysis of rough-grunts produced by wild chimpanzees at Gombe National Park, Tanzania, examining acoustic variation in call duration and peak frequency both within and between feeding bouts. We found that peak frequency, but not duration, displayed a bimodal distribution, supporting the view that rough-grunts include at least two acoustic sub-types. Nevertheless, calling bouts produced within each feeding bout reliably encompassed this full range of acoustic variation and calls did not display consistent temporal patterns of variation throughout feeding bouts. Our findings are thus inconsistent with the idea that rough-grunts label food properties. Rather, we argue that rough-grunts broadcast information about the signaler's foraging or social intentions.

11:20

2aSC8. Michael Owren's contributions to methods and models of vocal production for human and non-human primates. Drew Rendall (Univ. of NB, Sir Howard Douglas Hall 320, Fredericton, NB E3B 5A3, Canada, d.rendall@unb.ca)

This part of the special session comprises a review of some of the highlights of Michael Owren's many contributions to adapting methods and models of vocal production across primate species, punctuated by comments and discussion by the invited speakers, in preparation for the larger panel discussion at the end of the session.

11:40–12:00 Panel Discussion

TUESDAY MORNING, 8 MAY 2018 GREENWAY H/I, 8:20 A.M. TO 10:05 A.M.

Session 2aSP

Signal Processing in Acoustics, Underwater Acoustics, Physical Acoustics, and Noise: Time Reversal Acoustics

Brian E. Anderson, Chair
N145 Esc, Brigham Young Univ., MS D446, Provo, UT 84602

Invited Papers

8:20

2aSP1. Localization of sources of noise in moving atmospheres using time reversal and geometric acoustics. 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)

Due to acoustic refraction and convection in moving, inhomogeneous atmospheres, the backazimuths of recorded signals can be significantly different from the source backazimuth, posing a significant difficulty in source localization. Although an inverse problem, localization of sources of infrasound in such atmospheres has mostly been made using the forward ray-theoretic model of sound propagation. This approach uses search grids for the unknown model parameters that are subsequently estimated by optimizing an objective function. This approach can give rise to a large computational time as well as to the issue of uniqueness, stability, and accuracy of the solutions. The current paper presents a technique which avoids these issues, and which combines time reversal and geometric acoustics for source localization in such atmospheres. This technique, which can also be useful for source localization of other forms of acoustic signals in refractive media, effectively transforms the inverse problem into a forward one by utilizing a flow reversal theorem in the ray tracing equations. Beside the atmospheric conditions, the only parameters needed in this technique are the trace velocities of the signals recorded by at least two arrays. Comparison of numerical simulations with observational data will be presented.

8:40

2aSP2. Synthetic time reversal and remote acoustic sensing of changes in a vibrating structure. David R. Dowling and Tyler J. Flynn (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109-2133, drd@umich.edu)

Synthetic time reversal (STR) is a method for blind deconvolution that is based on the time-reversal symmetry of acoustic fields. It can be used to estimate an unknown acoustic source's broadcast waveform in an unknown multipath environment from remote transducer array recordings. This presentation will review the formulation of STR, and its use in ocean acoustics and non-destructive evaluation. In particular, experimental results are presented for the remote acoustic detection of mechanical changes to a vibrating 0.3-m-square by 3-mm-thick aluminum plate in a reverberant environment. The plate has nominally clamped edges and is subject to swept-frequency base excitation from 100 to 2,000 Hz. Sound radiated from the plate with and without a mechanical change is recorded remotely with a 15-microphone linear array and processed using STR to reconstruct a single radiated-sound signal that is deconvolved from the environment's unknown reverberation response at signal-to-reverberation ratio levels of -7 to -13 dB. The corrected signals are processed to detect boundary clamping defects and various length cuts in the plate via standard statistical baseline comparison techniques. Detection results using STR are superior and more robust to geometrical uncertainties than equivalent results from conventional approaches. [Sponsored by ONR and NAVSEA through the NEEC]

9:00

2aSP3. Multiple time-reversal focusing with a virtual source array. Gihoon Byun (Korea Maritime and Ocean Univ., Korea Maritime Univ., Dongsam 2-dong, Yeongdo-gu, Busan 606-791, South Korea, gihoonbyun77@gmail.com), H. C. Song (Scripps Inst. of Oceanogr., San Diego, CA), and J. S. Kim (Korea Maritime and Ocean Univ., Busan, South Korea)

Time reversal (TR) is the process of generating a spatio-temporal focus at a probe source (PS) location by backpropagating a time-reversed version of a received signal. While TR focusing requires the PS for a coherent acoustic focus at its origin, the requirement of the PS has been partially relaxed by the concept of a virtual source array (VSA) [J. Acoust. Soc. Am. 125, 3828–3834 (2009)]. A VSA can serve as a remote platform or lens and redirect a focused field to a selected location beyond the VSA for which the field is assumed as a homogeneous medium with constant sound speed. The objective of this study is to extend VSA-based single TR focusing to simultaneous multiple focusing. This is achieved using the optimization theory by employing the multiple constraints method derived from a constraint matrix, which consists of appropriately synchronized transfer functions. It is found that simultaneous multiple focusing can be achieved with distortionless response at selected multiple locations, and its performance degrades in the presence of sound speed mismatch. Possible applications for the VSA-based TR focusing are discussed, and numerical simulation results are presented

Contributed Papers

9:20

2aSP4. Time reversed elastic nonlinearity diagnostic applied to stress corrosion cracking in stainless steel. Sarah M. Young, Brian E. Anderson, and Stephen M. Hogg (Phys. and Astronomy, Brigham Young Univ., ESC N203, Provo, UT 84601, sarahmyoung24@gmail.com)

The Time Reversed Elastic Nonlinearity Diagnostic (TREND) is an inspection technique that enables one to locate damage in a structure using time reversal processing. TREND has been used to detect various types of damage in metals and rocks, but little has been done specific to stress corrosion cracking (SCC) other than the work of Anderson *et al.* J. Acoust. Soc. Am. **141**, EL76–EL81 (2017). SCC in stainless steel, especially around welds, is produced as a result of elevated temperature and saline conditions, and so has different properties from fatigue cracking or delaminations, both of which have been studied with TREND. In current research, welded steel samples exposed to a magnesium chloride bath were created. A number of samples were exposed for successively longer durations in order to study the corrosion damage level versus exposure time. Using TREND, samples were scanned then evaluated for indicators of local cracking at each scan point. Specific changes in spectral characteristics that are indicators of cracking were found to correlate to a longer exposure time and therefore the extent of SCC damage at that location.

9:35

2aSP5. Understanding the effect of room parameters on acoustic time reversal. Michael Denison and Brian E. Anderson (Phys., Brigham Young Univ., 485 S State St, Apt. 306, Provo, UT 84606, michael.denison23@gmail.com)

Time Reversal (TR) is a technique used to focus an acoustic signal at a particular point in space. Of the many variables that contribute to the quality of TR focusing, the most important are the number of sound sources, signal bandwidth. and properties of the medium. Although much research has been done to quantify the effects that the number of sound sources and signal bandwidth have on the TR process, little has been done in regard to the effect that room parameters have on TR. This is largely due to the difficulty involved with changing room acoustic parameters to measure their effects. We use the image source method (using the algorithm proposed by Allen and Berkley) to simulate the TR process in a variety of rooms with different acoustic and geometric room parameters. We define and calculate the maximum focal amplitude, the temporal focus quality and the spatial focus quality for each simulation. We compare the results and determine the effects of absorption and room volume on TR. We find that less absorption improves max response and spatial quality while it decreases temporal quality and that larger volumes have decreased max response and spatial quality while having increased temporal quality.

9:50

2aSP6. Using time reversal focusing for active control of magnetic resonance imaging noise. Trent Furlong and Brian E. Anderson (Dept. of Phys. & Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, trentfurlong@gmail.com)

The noise produced by a Magnetic Resonance Imaging (MRI) scan can cause discomfort to patients. Active Noise Control (ANC) systems have been developed to help reduce the amount of noise a patient experiences. However, due to the small enclosure of the MRI machine, ANC equipment can be cumbersome to use, and is limited to non-ferrous material. Time Reversal Acoustics (TRA) is a technique that can focus sound to a selected position in space that may be far from actual sound sources. We are exploring using TRA to deliver an ANC signal by broadcasting "anti-noise" signals that are focused to the patient's ears through TRA to cancel the MRI noise. This presentation will explore the applicability of using TRA to deliver different types of ANC signals to determine limitations in this process and show some proof of concept results.

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, 7 May 2018, from 5:00 p.m. to 6:00 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, 8 May 2018	11:00 a.m.–12:15 p.m.	S12, Noise
Tuesday, 8 May 2018	2:00 p.m.–3:00 p.m.	ASC S3/SC 1, Animal Bioacoustics
Tuesday, 8 May 2018	3:15 p.m.–4:30 p.m.	ASC S3, Bioacoustics
Tuesday, 8 May 2018	4:45 p.m.–5:45 p.m.	ASC S1, Acoustics

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 S12	**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
M. A. Bahtiarian, Chair ASC S12	ISO/TC 43/SC 3, Underwater acoustics	ASC S1, ASC S3/SC 1 and
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. J. Evans, Chair	ISO/TC 108/SC3 Use and calibration of vibration and shock measuring instruments	ASC S2
D. D. Reynolds, Chair	ISO/TC 108/SC4 Human exposure to mechanical vibration and shock	ASC S2 and ASC S3
D. J. Vendittis, Chair	ISO/TC 108/SC5 Condition monitoring and diagnostics of machine systems	ASC S2
IEC		
D. A. Preves and C. Walber, U.S. Technical Co-advisors	IEC/TC 29 Electroacoustics	ASC S1 and ASC S3

Meeting of Accredited Standards Committee (ASC) S12 Noise

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

D. F. Winker, Vice Chair ASC S12
ETS-Lindgren Acoustic Systems, 1301 Arrow Point Drive, 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, 8 May 2018.

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 AFTERNOON, 8 MAY 2018 NICOLLET C, 1:00 P.M. TO 4:40 P.M.

Session 2pAA

Architectural Acoustics: Interactions Between Acoustics and Architectural Design

Ana M. Jaramillo, Cochair
Ahnert Feistel Media Group, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444

Adel Hinawi, Cochair
Acoustic Distinctions, One Grand Central Place, 60 East 42nd Street, Suite 2036, New York, NY 10165

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. The architect as Ally: A multi-sensory approach to design. Gregory A. Miller and Robin S. Glosemeyer Petrone (Threshold Acoust., LLC, 141 W. Jackson Boulevard, Ste. 2080, Chicago, IL 60604, gmiller@thresholdacoustics.com)

The end product of an architectural acoustic design is not the quality of the documentation or the successful achievement of a set of parameters, but rather the experience of the end users. With rare exception, the perception of acoustic characteristics cannot be separated from the users' visual and tactile experiences. To unify these sensory experiences, we have adopted an approach to acoustic design that fully embraces the architect's spatial and material vision for a building. At the same time, we seek to educate the architect so that they can embrace and internalize the acoustic goals of the building in their work. This presentation will describe our overall design approach, means of communication and demonstration used to build conviviality with our clients, and examples where acoustic and architectural goals have coincided.

1:25

2pAA2. Successfully persuading architects' form to follow acoustical function. David A. Conant (5655 Lindero Cyn Rd., #325, McKay Conant Hoover Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, dconant@MCHinc.com)

Architect Louis Sullivan's mantra, *Form follows function,* is no better embedded in any subdiscipline of architecture than that of building acoustics, and this is especially true in the realm of performing arts. This paper describes several examples of performing arts spaces in which MCH involved architects, often rather directly, into acoustical design. These studies were characterized, not by computer-centric tools, but rather were often most effectively and intuitively realized through simple light models that informed optimal room shaping real-time, hand-on exercises with our clients. Some solutions, especially appreciated by architects, successfully invoked indigenous or traditional forms that lent themselves beautifully to excellent speech intelligibility, musical clarity, and envelopment. This paper will discuss projects with acoustically-effective forms drawn from Native American kivas and Taoism, to Moorish tessellated patterns, to geographic forms such as southwest canyons and quarries, and more—including even "hanging chads."

1:45

2pAA3. Acoustical conflicts and synergies with energy efficient building design. Ralph T. Muehleisen (Energy Systems, Argonne National Lab., 9700 S. Cass Ave., Bldg 221, Lemont, IL 60439, rmuehleisen@anl.gov)

In the US, commercial and residential buildings consume more than 40% of the total energy and more than 75% of all electricity, contributing greatly to carbon emissions. In order to reduce both carbon emissions and reduce utility costs, buildings are being designed and constructed with much higher efficiency than even just a decade ago. Indeed, a "net-zero-ready" building is both possible and even affordable. Some design strategies can also improve building acoustics but others are detrimental. In this talk, synergies and conflicts of acoustics and energy efficient design are discussed. In particular, well-sealed building envelopes and use of high performance windows can provide both acoustic and energy benefits. However, the use of natural ventilation is at odds with acoustics. In this talk, the conflicts and synergies of these high efficiency building treatments are presented.

2:05

2pAA4. Acoustical balance between the stage and the pit in the Teatro Colón of Buenos Aires. Gustavo J. Basso (Facultad de Bellas Artes, Universidad Nacional de La Plata, Argentina, Calle 5 N° 84, La Plata, Buenos Aires 1900, Argentina, gustavobasso2004@yahoo.com.ar)

The acoustical balance between the singers and the orchestra in the well-known Teatro Colón of Buenos Aires, as an opera theater, is evaluated from several perspectives. The almost ideal balance seems to be a result of a particular combination of architectural features: among others, the shape of the horseshoe, the height and depth of the boxes at the upper levels, and the design of the proscenium and pit. In this sense, the reflections on the stage floor become significant for the higher levels. Some of the architectural causes of this acoustical behaviour have been found out from the results of opinion polls on the perceived sound by the audience, physical measurements, and the outcomes of a digital model. This paper analyses the balance between the singers, placed in various locations on the stage, and the orchestra in the pit. As it will be seen, the architectural characteristics of the theater allows the musicians to maintain the balance in almost ideal values and to preserve the spectral equilibrium and bass response through the entire hall.

2:25

2pAA5. Creative collaborations in the design of buildings. Gary W. Siebein, Hyun Paek, Marylin Roa, Keely Siebein, Jennifer R. Miller, Matthew Vetterick, and Gary Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

Integrating acoustical design features in rooms that fulfill acoustical, architectural, mechanical, and other functions simultaneously requires close collaboration among design team members. The efforts involved each team member understanding the other disciplines to the extent that they could creatively interpret the work. Three case studies will highlight the advantages of this type of collaboration. Full size mock-ups, computer modeling, auralizations, and other advanced design tools assist in this process. The first is a wall panel in a large museum that is used as a spatial divider, a sound attenuating return air plenum, and a surface for exhibiting artwork. A multi-function atrium was used as the central organizing feature in another museum. Design efforts explored how sounds propagated from one level to the next could be reduced. Multi-functional, layered wall assemblies that contained variable acoustic features, sound diffusing/reflecting panels, supply and return air distribution and layers to reduce sounds entering the building from the exterior were used in the renovation of a college theater. Multiple iterations of acoustical, architectural, mechanical and interior design efforts followed by intense analysis and discussion proved very rewarding for the design team with beautiful looking and sounding spaces.

2:45

2pAA6. Balancing uniform visual appearance with the need for adjustable acoustics in a concert venue: A case study. David Kahn (Acoust. Distinctions, Stamford, CT), Joshua R. Dunham (Acoust. Distinctions, 60 East 42nd St., New York, NY 10165, jdunham@ad-ny.com), Alexander Aquila, Tyler Cottrell, Jacob Ott, Nicholas Roselli, and Robert Celmer (Acoust. Prog. and Lab., Mech. Eng. Dept., Univ. of Hartford, West Hartford, CT)

An acoustician's recommendations for adjustable acoustics curtains in a concert venue sometimes comes into conflict with the architect's aesthetic vision for the space. The architectural intent is usually for patrons to have a uniform visual experience of the room, regardless of the setting of the adjustable acoustics elements. This leads to the need for architecturally interesting, acoustically transparent elements, such as perforated metal or wooden grills, that can fully or partially hide the adjustable curtains from the eyes of the audience. This paper presents a summary of the collaboration between Acoustic Distinctions and the architectural firm HGA in designing sound-transparent, patterned wood grills that enable a visually attractive and consistent architectural aesthetic, regardless of the settings of the adjustable acoustics curtains behind, in the Kracum Performance Hall at Carleton College. The wood grills—designed collaboratively by the acoustician and architect—were mocked-up and acoustical tests were carried out at the University of Hartford to document their acoustical transparency. Results from those tests will be presented. These results may provide helpful guidelines to future collaborations with architects and acousticians on the design of acoustically transparent surfaces.

3:05–3:20 Break

3:20

2pAA7. New sound system design in concert halls. Wolfgang Ahnert (ADA Acoust. & Media Consultants GmbH, Arkonastr. 45-49, Berlin D-13189, Germany, wahnert@ada-amc.eu)

In 2017, two concert halls have opened in Germany, in January Hamburg's Elb-Philharmonic Hall and in Dresden the concert hall inside the existing Kulturpalast. The presentation will report briefly about the history and development of both projects. The sound systems in both halls must serve as an announcement system but also in case of emergency as a Speech alarm system according to the German standard DIN VDE 0833-4. For both halls, the target values of Speech Intelligibility have been developed and the design approach for the needed sound systems is explained in the presentation, not only for the halls but for the partially complicated lobbies too. The excellent architectural design of the two halls led to many problems to ensure the required performance of the speaker systems, but to hide their physical visibility. An influential issue was here the interaction with the existing room-acoustic parameters in both facilities. With high reverberation values and expected high noise floors in case of emergencies the limits of needed speech intelligibility had been reached very fast. This interaction will be discussed and simulation and some measurement results are given.

Contributed Papers

3:40

2pAA8. Development and implementation of a construction noise and vibration management plan for occupied healthcare facilities. Gina Jarta and Elliott Dick (HDR, 701 Xenia Ave. South, Ste. 600, Minneapolis, MN 55416, Gina.Jarta@hdrinc.com)

Acoustics and noise management play an integral role in providing an optimum care environment for patients and staff of healthcare facilities. Indoor noise and vibration affect patient comfort and healing, clear communication, operation of sensitive imaging equipment, and overall occupant satisfaction. Construction and demolition activities often occur in occupied hospitals requiring ongoing noise and vibration sensitive operations to remain active. This case study presents the development and implementation of a construction noise and vibration management plan for a neonatal intensive care unit (NICU) expansion project. The expansion included the addition of 23 NICU beds an existing roof level, directly adjacent to an existing NICU unit, Post-Anesthesia Care Unit, operating rooms, and *In Vitro* Fertilization clinic which remained in operation during demolition and construction. The acoustical design process from establishment of baseline conditions, prediction of project-related noise and vibration, design of mitigation strategies, and development of the management plan will be presented. Monitoring data collected over the 15 month demolition and construction period will be reviewed along with project implementations strategies and lessons learned.

3:55

2pAA9. Comparative analysis of resilient and dense materials in lightweight construction for impact sound attenuation. Sean Harkin, Jacob Watrous (Eng., SoundSense, LLC, PO Box 1360, Wainscott, NY 11975, sean@soundsense.com), Bonnie Schnitta (Eng., SoundSense, LLC, East Hampton, NY), and Jennifer Scinto (Eng., SoundSense, LLC, Wainscott, NY)

Resilient underlayments are commonly utilized as a primary method of reducing footfall noise in architectural acoustics. Although resiliency is a large component of a partition's footfall performance, as well its ability to achieve higher AIIC ratings through ASTM E1007 AIIC testing, adding resiliency alone does not always address all of the frequencies which cause disturbances due to footfall. Particularly in lightweight construction, the density of the configuration is also a critical component of a successful solution. Due to the lack of sufficient mass in lightweight construction materials, successful treatment for impact noise disturbances in lightweight conditions becomes more difficult to achieve. This paper compares data with different flooring configurations in mock-up ASTM AIIC testing conditions in order to evaluate advantages and disadvantages of ANISPL (Absorption Normalized Impact Sound Pressure Level) performance in different frequencies. Resiliency and density are added in varying combinations in a wood frame construction in order to better understand their relationship to AIIC ratings and a partition's success in impact noise insulation.

4:10–4:40 Panel Discussion

Session 2pAB

Animal Bioacoustics: Plant Bioacoustics

Aaron Thode, Cochair
SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238

Simon E. Freeman, Cochair
Scripps Institution of Oceanography, 7038 Old Brentford Road, Alexandria, VA 22310

Invited Papers

1:20

2pAB1. Insect sound production and transmission in plant materials of different compositions and structures. Richard Mankin (Ctr. for Medical Agricultural and Veterinary Entomology, USDA ARS, USDA ARS CMAVE, 1700 SW 23rd Dr., Gainesville, FL 32608, Richard.Mankin@ars.usda.gov)

Insects use plants for food and shelter, and many species also have taken advantage of plant acoustical and structural characteristics to communicate for mating and social interaction over extended distances without expending significant energy. Humans have taken advantage of plant acoustical and structural characteristics to detect hidden insect infestations passively by monitoring their feeding and movement activities. This presentation reviews the characteristics of sound transmission as well as the characteristics of insect movement and feeding sounds in plant structures and products. Although insect sounds can be masked by loud background noise, different species produce sounds with particular spectral and temporal patterns that help distinguish them from background signals and from each other. Several practical applications of insect bioacoustics are discussed, including disruption of insect mating, targeting of tree pests, and monitoring of the time course of different pest management treatments.

1:40

2pAB2. Sound of wood-boring larvae and its automated detection. Alexander Sutin, Alexander Yakubovskiy, Hady Salloum, Timothy Flynn, Nikolay Sedunov (Stevens Inst. o Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu), Hannah Nadel, and Sindhu Krishnankutty (PPQ S&T, USDA APHIS, Buzzards Bay, MA)

Stevens Institute of Technology has been investigating solutions for instrumental detection of invasive species at ports of entry. Stevens has built several acoustic systems for detection of acoustic/ vibrational signals produced by insects. This paper presents acoustic signals recorded in tests conducted in APHIS Otis Lab using tree bolts infested by Asian Longhorn Beetle, ALB and Emerald Ash Borer, EAB larvae. The analysis of the recorded sounds extracted the signal features that allowed larval classification. These features include frequencies of the generated pulses, their durations and frequencies of pulse envelops. These features showed a clear separation of ALB and EAB. For example, the main frequency of the ALB sound was in the range of 3.8–4.8 kHz, while for EAB it was between 1.2 and 1.8 kHz. A preliminary algorithm for automated insect signal detection was developed. The algorithm automatically detects pulses with parameters typical for the larva-induced sounds and rejects non-insect sound pulses. Detection is announced when the number of detected pulses for some time (5 min) exceeds the definite threshold. In the conducted test, this algorithm provided detection of a larva in all tested samples without false alarms. [This project was funded under contract with the U.S Department of Homeland Security (DHS) Science and Technology Directorate (S&T), contract HSHQDC-10-A-BOA35.]

2:00

2pAB3. Acoustic interactions between plants and animals. Michael G. Schöner and Caroline R. Schöner (Appl. Zoology and Nature Conservation, Univ. of Greifswald, Loitzer Strasse 26, Greifswald 17121, Germany, schoenerm@uni-greifswald.de)

Acoustic communication and reactions to acoustic cues are widespread and intensively studied in animals but have largely been neglected in other organisms such as plants. However, there is growing evidence for acoustic communication in plant-animal interactions. While knowledge about active acoustic signaling in plants (i.e. active sound production) is still in its infancy, research on passive acoustic signaling (i.e. reflection of animal sounds) revealed that bat-dependent plants have adapted to the bats' echolocation systems by providing acoustic reflectors, which attract mutualistic animal partners. Studies also show that plants are able to perceive sound and thus, potentially can react to animals (e.g., physiologically). Moreover, in the course of evolution plants should become acoustically more attractive to mutualistic animals that find their plant partners based on sound and less conspicuous to parasites. The current challenge is to discover further examples of plants and animals that acoustically interact with each other. Understanding the underlying proximate mechanisms and ultimate causes of acoustic communication will shed light on an underestimated dimension of information transfer between plants and animals.

2:20

2pAB4. Ultrasonic Transmission Behavior in *Posidonia oceanica* Rhizomes. Jay R. Johnson (Mech. Eng. Dept. and App. Res. Labs., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, johnson.jayrichard@utexas.edu), Jean-Pierre Hermand (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, Brussels, Brussels Capital, Belgium), and Preston S. Wilson (Mech. Eng. Dept. and App. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The roots and rhizomes of seagrass can form a complicated multi-phase layer within sediments. Gas channels in the rhizomes, known as aerenchyma, give rise to a complicated acoustic response. A model for acoustic propagation through such rhizomes would be beneficial for acoustic remote sensing communities. *Ex situ* measurements of the ultrasonic sound speed through the dense woody structure of the rhizomes of the seagrass *Posidonia oceanica* are presented. Ultrasonic (1–5 MHz) time-of-flight measurements were made with the rhizomes segments aligned both lengthwise and crosswise to the propagation direction of the acoustic pulse. The acoustic behaviors of plants collected from different locations in the Mediterranean are compared to each other and to the behaviors of the associated leaf blade tissues. Two measurements were also made of rhizomes before and after degassing the aerenchyma, to quantify the effects of entrained gasses on acoustic behavior. [Work supported by ONR and ONR Global.]

2:40

2pAB5. Acoustics of seagrass photosynthesis. Jean-Pierre Hermand (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, av. F.D. Roosevelt 50, CP165/57, Brussels, Brussels Capital 1050, Belgium, jhermand@ulb.ac.be), Jay R. Johnson (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, Austin, TX), Olivier Debeir (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, Brussels, Belgium), and Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, Austin, TX)

The paper reviews experiments in the Mediterranean that have investigated the use of low frequency sound transmission measurements (0.2–20 kHz) to study metabolism of seagrasses *in situ* in a noninvasive way and to better evaluate primary production at the scale of a meadow, for *Posidonia oceanica and Cymodocea nodosa* species. As sound interacts with the canopy and rhizosphere, the cumulative effect of multiple scattering from the aerenchymatic tissues of leaf blade, rhizome, and root changes the character of the environment impulse response. Diel variations of received energy and statistical features are due to oxygen movement which modifies the plant scattering function. During periods of high net photosynthesis, pressurization of the lacunal gas modifies the tissue mechanical properties. High internal oxygen partial pressures cause the release of oxygen into the diffusive boundary layer which can result in bubble formation on the tissue surface when local oxygen concentration exceeds solubility. *In situ* experimental results will be discussed in the light of laboratory acoustic and microscopic investigations of the tissues. [Work supported by ONR, ONR Global.]

3:00

2pAB6. Acoustic emissions by marine algae during photosynthesis. Simon E. Freeman (Naval Undersea Warfare Ctr., 6819 Duke Dr., Alexandria, VA 22307, simon.freeman@gmail.com), Lauren Freeman (Naval Undersea Warfare Ctr., Washington, District of Columbia), Giacomo Giorli (National Inst. of Water and Atmospheric Res., Honolulu, Hawaii), and Andreas F. Haas (NIOZ Royal Netherlands Inst. for Sea Res. and Utrecht Univ., Amsterdam, Netherlands)

Coastal underwater soundscapes typically contain signals from soniferous, biological processes. Identifying the sources that contribute to oftentimes acoustically complex soundscapes would facilitate a noninvasive, remote, and volumetrically integrative method to survey underwater ecological state, as has been demonstrated on land. To date the lack of knowledge on these sources has posed a challenge for extracting meaningful ecological information from underwater biological soundscape recordings. We have observed that photosynthesis by macroalgae is an acoustically active process, driven by oxygen bubbles separating from the algal surface and ringing at the Minnaert frequency. The resultant soundscapes correlate with benthic macroalgal cover across ecological gradients on shallow Hawaiian reefs during periods of daylight. Bubble size, production rate, and sound exposure level follow the concentration of dissolved oxygen, which in turn rises and falls with the availability of photosynthetically active radiation. Increased macroalgal cover results from climate related stress, overfishing and/or pollution through nutrient runoff and serves as an indicator of degradation in many underwater ecosystems. Our observation lays the foundation for signal processing methods that quickly and passively evaluate relative algal abundance in littoral waters. These findings can also be applied towards more accurate monitoring of primary production in industrial processes.

3:20–3:35 Break

Contributed Papers

3:35

2pAB7. Mating vibrational signal transmission through and between plants of an agricultural pest, the Glassy-Winged Sharpshooter. Shira D. Gordon (U.S. Dept. of Agriculture, 9611 Riverbend Ave., Parlier, CA 93648, shira.gordon@ars.usda.gov), Benjamin Tiller, James F. Windmill (Ctr. for Ultrasonic Eng., EEE, Univ. of Strathclyde, Glasgow, United Kingdom), Peter M. Narins (Dept. of Integrative Biology & Physiol., UCLA, Los Angeles, CA), and Rodrigo Krugner (U.S. Dept. of Agriculture, Parlier, CA)

The agricultural pest, glassy-winged sharpshooter (GWSS), *Homalodisca vitripennis*, relies primarily on successful vibrational communication across its home plant. Males and females engage in a vibrational duet to identify correct species, attractiveness of mate, and location on the plant. The signal produced by these animals has a dominant frequency component between 80 and 120 Hz, with harmonics spaced approximately 100 Hz apart. However, our analysis revealed that not all harmonics are present in every recorded signal. Therefore, we sought to understand how the GWSS vibrational communication signal changes over distance on the plant. We have confirmed that first, with increasing distance fewer high frequency harmonics are present. Second, at distances of only 50 cm, there is a difference in the latency of signal arrival based on the frequency, with higher frequencies arriving sooner. Finally, the animal appears to generate no airborne signal component, yet, the low frequencies are clearly detectable in neighboring plants by the signal "jumping" from leaf-to-air-to-leaf. Together, these results highlight the complexity of vibration transmission in plants and the possibility of alteration and disruption of the GWSS signal.

3:50

2pAB8. Acoustic sensing of macroalgae (kelp) for large-scale marine biomass production. Erin M. Fischell (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 77 Massachusetts Ave., 5-204, Cambridge, MA 02139, emf43@mit.edu), Andone C. Lavery, Amy Kukulya, and Timothy K. Stanton (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

An important component of the future use of large-scale offshore farms to grow macroalgae (kelp) will be remote monitoring of infrastructure, the environment, and plant health over areas so large that manual inspection is not practical. A new program, the Advanced Research Projects Agency-Energy's Macroalgae Research Inspiring Novel Energy Resources (ARPA-E MARINER), has the long-term goal of domestic energy production using biofuel derived from macroalgae. As part of that program, an integrated sensing system is being developed for unmanned underwater vehicle (UUV) monitoring of infrastructure, macroalgae growth, water properties, and associated organisms in experimental offshore macroalgae farms occupying areas square kilometers in size. A critical component of this monitoring system is acoustic sensing using a split-beam sonar system. Time-of-flight and volume backscattering data from the echosounder will be used to determine the thickness of growth and percentage volume inhabited of macroalgae. The objective is to provide a map correlated to biomass distribution variability across the farm area. Data will also be collected on aggregations of fish and zooplankton both within and outside the farm. Early results from local tests on sugar kelp will be presented, including initial research into the correlation between these acoustic data and biomass.

4:05

2pAB9. Effect of carbon content on sound speed and attenuation of sediments in seagrass meadows. Gabriel R. Venegas (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712-1591, gvenegas@utexas.edu), Aslan Aslan, Ivy M. Hinson, Abdullah F. Rahman (School of Earth, Environ., and Marine Sci., The Univ. of Texas Rio Grand Valley, Brownsville, TX), Kevin M. Lee, Megan S. Ballard, Jason D. Sagers, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Justin T. Dubin, and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Globally, seagrass-bearing sediments contain 19.9 billion metric tons of carbon (C), and account for 10% of all organic C buried in the ocean each year. Protection of these C stores is vital in mitigating climate change [Fourqurean, J. W., *et al.,* Nature Geoscience 5, 505–509 (2012)]. Some sediment acoustic properties are sensitive to the presence of gas bubbles entrained in such C stores due to inherent anaerobic decomposition. Measurement of these properties could therefore provide a means to indirectly monitor C stores and overall seagrass meadow productivity. As a preliminary effort to investigate the relationship between C content and acoustic properties of seagrass-bearing sediments, cores were collected in the seagrass meadows of Lower Laguna Madre, Texas. Sound speed and attenuation from 100 kHz to 300 kHz were measured radially in 2-cm-depth increments. The cores were subsequently frozen, sliced along the same depth increments, and their C content estimated using an elemental analyzer. Acoustic properties were compared with C content along the depth of each core. The potential for sound speed and attenuation of seagrass-bearing sediments to be used as a metric for monitoring seagrass meadow productivity will be discussed. [Work supported by ONR and ARL:UT IR&D.]

TUESDAY AFTERNOON, 8 MAY 2018 GREENWAY A, 1:00 P.M. TO 4:40 P.M.

Session 2pAO

Acoustical Oceanography and Underwater Acoustics: Acoustic Seabed Characterization II

David P. Knobles, Cochair
none, KSA LLC, PO Box 27200, Austin, TX 78755

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

Invited Papers

1:00

2pAO1. Quantifying the effect of random roughness on synthetic aperture sonar image statistics. Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, University of New Hampshire, Durham, NH 03824, anthony.lyons@ccom.unh.edu), Derek R. Olson (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), and Roy E. Hansen (Norwegian Defence Res. Establishment (FFI), Kjeller, Norway)

A perturbation-theory-based model has been developed to predict the effect of random seafloor roughness on synthetic aperture sonar (SAS) image statistics. The continuous variation in scattering strength produced by a random slope field is treated as an intensity scaling on image speckle produced by the SAS imaging process. Changes in image statistics caused by roughness are quantified in terms of the scintillation index (SI). Factors influencing the SI include slope variance, geo-acoustic properties of the seafloor, the probability density function describing the speckle and the signal-to-noise ratio. Example model-data comparisons will be shown for SAS images taken off the coast of Tellaro, Italy, by the NATO Undersea Research Centre, La Spezia, Italy (now the NATO Centre for Maritime Research and

Experimentation) using the 300 kHz MUSCLE SAS system, and for data collected with the Norwegian Defence Research Establishment's 100 kHz HISAS system off the coast of northwest coast of Elba Island, Italy, in 2013. Parameter estimates obtained from high-resolution SAS data collected in these experiments will be used to illustrate the efficacy of the model and its possible use for estimating roughness parameters, such as root-mean-square slope. [Work performed under ONR Grant Nos. N00014-10-1-0051, N00014-10-1-0047, and N00014-10-1-0151, and NOAA Grant No. NA15NOS4000200.]

1:20

2pAO2. Geacoustic inversion using autonomous underwater vehicles. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu), Venugopalan Pallayil (National Univ. of Singapore, Singapore, Singapore), and Erin Fischell (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Autonomous underwater vehicles (AUV's) are valuable for bottom geoacoustic studies because they can remain in close proximity to the seafloor, maneuver accurately along underwater tracks, and perform multi-platform operations. A REMUS 100 AUV was deployed in the recent Seabed Characterization Experiment, sponsored by the Office of Naval Research (ONR) and conducted in the New England Mud Patch Area from March to April 2017. This AUV was equipped with a sound source transmitting acoustic chirp signals from 800 to 1300 Hz and a 20 m-long digital thin line towed array (DTLA) receiving both the AUV source signals and acoustic signals transmitted from different sources. The AUV track was along a path between a moored source and a fixed receiver sled. With this experimental configuration, three different geoacoustic approaches were investigated: (1) bottom reflection inversion using the AUV source and the towed DTLA, (2) range dependent broadband acoustic inversion using the moving AUV source and the fixed receiver, and (3) range average broadband inversion using the moored source and the fixed receiver. To keep a precise time base, Chip-Scale Atomic Clocks (CSAC's) were utilized in the AUV source, the moored source and the fixed receiver, and the DTLA is synchronized to the AUV source via high-resolution acoustic interrogations. The inversion results will be presented, along with discussions for future work. [Work supported by ONR and ONRG.]

1:40

2pAO3. Estimation of sound speed and attenuation in muddy sediments using spatial coherence measurements of sound propagation. Lin Wan and Mohsen Badiey (Univ. of Delaware, 104 Robinson Hall, Newark, DE 19716, wan@udel.edu)

In the spring of 2017, a multi-national and multi-institute shallow water propagation experiment was conducted in the New England Mud Patch in order to study the acoustic properties in muddy sediments. Different types of acoustic signals (i.e., 31-g explosive charges, combustive sound source, multi-tone, and linear frequency modulation signals) deployed at various ranges, depths and azimuths were measured by one L-shaped array, one horizontal line array, and several vertical line arrays within the 30 km $\times$ 10 km experimental area. In this paper, these measured signals are first utilized to obtain the correlation coefficients for vertical coherence (VC) and longitudinal horizontal coherence (LHC), which are sensitive to the seabed geo-acoustic parameters. These spatial coherence measurements are then applied in the VC and LHC based geo-acoustic inversion algorithms [Wan *et. al.* JASA, 2016] to infer sound speed and attenuation in muddy sediments. Finally, the results from VC and LHC based geo-acoustic inversion approaches are compared with those obtained by using normal mode characteristics (e.g., modal dispersive curve with Airy phase structure, modal attenuation coefficient, and mode shapes). [Work supported by ONR.]

2:00

2pAO4. Effects of azimuthal dependent sediment layer structure on broadband acoustic propagation during Seabed Characterization Experiment in 2017. Mohsen Badiey, Lin Wan (Univ. of Delaware, University of Delaware, Newark, DE 19716, badiey@udel.edu), and John A. Goff (Inst. for Geophys., Univ. of Texas, Austin, TX)

Seabed physical properties profoundly affect acoustic energy upon interaction. The inherent structural composition of the sediment column interacting with acoustic waves has been a topic of research for decades. Besides the intrinsic physical properties that cause signal attenuation, one of the features that various sediment structures share when interacting with acoustic signals, is causing azimuthal dependence on the broadband acoustic wave propagation. This phenomenon studied in late 1990's [Badiey *et al.*, JASA, 1997a, b] was revisited during the Seabed Characterization Experiment in 2017 (SBCE 2017). The sediment layer structure was obtained from the Chirp Sonar survey collected in the summer of 2015 during pilot study. These data plus CTD and bathymetric measurements are used to construct environmental input along radial tracks for acoustic field computations and range-varying wave number spectral calculations to support SBCE 2017 data analysis. The simulation results show azimuthal variability of acoustic normal modes similar to the experimental data documented in earlier studies [Badiey *et al.* JASA, 1997 and 2017]. Current results indicate that azimuthal variability of sediment physical properties is one of the causes of acoustic variability in this region. [Work supported by ONR 321OA.]

2:20

2pAO5. Ship azimuth prediction using supervised machine learning. Emma Reeves and Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, ecreeves@ucsd.edu)

Machine learning methods are applied to noise from the R/V Endeavor across several days during the SCEx17 experiment to predict the ship's azimuth. Sample Covariance Matrices (SCMs) are formed from received pressures on two vertical line arrays (VLA1, VLA2) and one horizontal line array (SWAMI). Previously, support vector machine (SVM), feed-forward neural network (FNN), and random forests (RF) machine learning methods have accurately estimated the range of a source towed in a linear geometry in shallow water (Niu *et al.*, JASA **142**, 1176–1188 (2017); Niu *et al.*, JASA **142**, EL455–460 (2017)) where the training and test tracks were close together in time. In this study, we investigate the robustness of SVM and FNN for circular track prediction when the training and test tracks are taken from different days with different sound speed profiles.

2:40–2:55 Break

Contributed Papers

2:55

2pAO6. Gradient-based Bayesian geoacoustic inversion for sediment properties at the New England mud patch. Josée Belcourt, Stan E. Dosso (Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Bob Wright Ctr. A405, Victoria, BC V8P 5C2, Canada, joseebelcourt@uvic.ca), Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA), and Jan Dettmer (Dept. of Geoscinece, Univ. of Calgary, Calgary, AB, Canada)

This paper presents nonlinear Bayesian inversion of wide-angle seabed reflection-coefficient data for fine-grained/cohesive sediments recorded in the ONR Seabed Characterization Experiment at the New England mud patch. The inversion is applied to high-resolution broadband reflectivity data from a site characterized by smooth bathymetry and a thick mud layer. Since smooth, continuous gradients in seabed properties are common for low-speed mud layers, this work develops a Bayesian inversion based on a Bernstein-polynomial representation of sediment geoacoustic profiles. The gradient-based Bernstein-polynomial inversion is compared to a trans-dimensional inversion which samples over profiles defined by an unknown number of discrete seabed layers. [The research was funded by the Office of Naval Research, Ocean Acoustics Program, and the Canadian Department of National Defence.]

3:10

2pAO7. End-fire synthetic aperture sonar for seafloor volume scattering studies. Shannon-Morgan M. Steele and Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, ssteele@ccom.unh.edu)

Acoustic returns from seafloor sediment are comprised of scattering from both the interface and sediment volume. Although volume scattering is often the dominant mechanism, direct measurements of this component have rarely been made, if at all, due to interface roughness biasing. This bias is especially prevalent at lower frequencies where beam widths are typically 30–40 degrees. Current synthetic aperture sonar (SAS) systems are side looking and achieve narrow beam widths by coherently combining multiple acoustic pings as the sonar moves. End-fire (forward-looking) SAS would formulate a synthetic array in the same direction of travel by vertically orienting a transducer and lowering it towards the seafloor while pinging. This would create a narrow beam, significantly reducing the interface roughness bias. End-fire SAS array gains are not as substantial as conventional side-looking SAS. However, beam pattern simulations suggest the gains are still significant: a synthetic array length of 100 wavelengths can reduce a sub-bottom profiler's 40 degree beam width to 7 degrees. This talk will discuss proof of concept, motion controlled experiments performed in an acoustic testing tank and in the field.

3:25

2pAO8. Pressure and particle velocity measurements from a broad band source at ranges 1–10 km. Peter H. Dahl and David R. Dall'Osto (Appl. Phys. Lab. and Mech. Eng., Univ. of Washington, 1013 NE 40th St, Seattle, WA 98105, dahl@apl.washington.edu)

The IVAR system (Intensity Vector Autonomous Recorder) is a bottom deployed system developed for first-use in the Sediment Characterization Experiment (SCE17), conducted on the New England Mud Patch [40°28' N, 70°35' W] in spring 2017. IVAR continuously and coherently records four channels of acoustic data, three from a tri-axial accelerometer embedded in a neutrally buoyant sphere (diameter 10 cm) and one from an omnidirectional hydrophone positioned 10 cm above the centroid of the sphere positioned 1.2 m above the seafloor. The connection of these measurements to understanding seabed properties as part of SCE17 has been discussed previously. Here, emphasis is placed on documenting pressure and particle velocity signals as these quantities evolve with range from a broad band explosive source (Mk64 SUS charge). We explore the phase relation between pressure and components of particle velocity through study of active (in phase) and reactive (out of phase) intensity, as well as corresponding non-dimensional indices of the acoustic vector field, and discuss how these are influenced by propagation conditions. The work has relevance to both the ongoing geoacoustic studies from SCE17 as well as to studies on the sensitivity of fish to acoustic particle velocity generated by explosions.

3:40

2pAO9. Estimation of parameters of gassy layer in sediment in shallow water using measurement of angular dependence and spectrum of reflection coefficient of wide-band signals. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 199 Abba Khouchy Ave., Mt. Carmel, Uni of Haifa, Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il), Andrey Lunkov (General Phys. Inst., Moscow, Russian Federation), Ernest Uzhansky (Marine GeoSci., Univ. of Haifa, Haifa, Israel), and Ilia Ostrovsky (IOLR, Kinneret Lab, Migdal, Israel)

Method of estimation of gassy layer parameters in sediment (sound speed and thickness) on the base of measurement of angular and frequency dependencies reflection coefficient of wideband signals is presented. Experiments were carried out in Lake Kinneret (Israel), which is characterized by remarkable organic content in sediment producing methane bubbles. Direct chemical, biological analysis and usage of frozen cores show comparatively narrow layer (a few tens of cm) changing in dependence on season. Principal point for acoustical method is low sound speed in bubble layer (up to 300–400 m/s) and big reflection coefficient from the corresponding half-space (0.6–0.7 for vertical incident). Due to narrow gassy layer, there should be half wavelength resonances at the corresponding frequencies. In experiments, wideband signals were used (500–2500 Hz) and receiving system: single hydrophone (1 m from source) and vertical line array. Estimation of layer's parameters was carried out using measured reflection coefficient as a function of frequency and angle of reflection. Resonance frequencies were 800 Hz and 1600 Hz. The corresponding analysis of data give values of thickness 25–30 cm, with the sound speed in layer 400–450 m/c. Results are compared with experimental data obtained by direct measurements. [Work was supported by ONRG and ISF.]

3:55

2pAO10. Estimating low-frequency sediment sound speed dispersion from the horizontal coherence of the head wave excited by a light helicopter. Dieter A. Bevans and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr, La Jolla, CA 92093-0238, dbevans@ucsd.edu)

A series of shallow-water acoustic experiments has been conducted off the coast of southern California using a Robinson R44 helicopter as a low-frequency ($\approx$ 13–3500 Hz) sound source. The aim of the experiments was to recover the sound speed of a fine to very-fine sand sediment from the horizontal coherence of the head wave excited in the water column by the helicopter. Two hydrophones, separated horizontally by approximately 15 m and situated 0.5 m above the seabed, received the head-wave signals, allowing the coherence function to be formed over the bandwidth of the airborne source. The sediment sound speed was recovered by matching the zero crossings of the measured coherence function to those predicted from a recently developed theory of head-wave generation in shallow water. Using this technique, the dispersion in the sediment sound speed can be estimated over a frequency range extending between 27 Hz, the lowest zero crossing of the coherence function, and 3.5 kHz, the bandwidth of the source. In the middle of the frequency band, the sound speed of the sediment was estimated to be 1682 $\pm$ 16 m/s, consistent with the known sediment type. [Research supported by ONR, SMART(DOD), NAVAIR, and SIO.]

4:10

2pAO11. Sound speed profiles in the global ocean calculated from WOCE data. Mukunda Acharya (Phys. and Astronomy, Univ. of MS, 112-, 114 Chucky Mullins Dr, Oxford, MS 38655, Oxford, MS 38655, mkachary@go.olemiss.edu) and Likun Zhang (Phys. and Astronomy, Univ. of MS, University, MS)

Sound speed in the stratified ocean varies continuously with the depth. The sound speed profile reaches a minimum at a depth, that is typically 1 km in mid-latitude to form a sound channel. In this study, the CTD data

collected in the World Ocean Circulation Experiment (WOCE) are used to calculate around 6,000 sound speed profiles over the global ocean. The data of conductivity, temperature and pressure at different depths are from measurements at intervals typically in 55 km by different cruises. Sound speed is examined from the data in each cruises. By analyzing sound speed profiles, sound speed minimum and the depth for this minimum are characterized as a function of latitude ranging from 70-degree south to 70-degree north. The results show a well determined pattern. Sound speed distributions over the global ocean at various depths in shallow water are also examined.

4:25

2pAO12. Characterization of soft marine sediments using vector sensors. Oleg A. Godin (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg 232, Monterey, CA 93943-5216, oagodin@nps.edu), Thomas J. Deal (Naval Undersea Warfare Ctr. Div. Newport, Newport, RI), and Hefeng Dong (Dept. of Electron. Systems, Norwegian Univ. of Sci. and Technol., Trondheim, Norway)

Horizontally and vertically polarized interface waves with phase speeds of 45–375 m/s were observed in the North Sea using a horizontal linear array of three-component seismometers [H. Dong *et al*., J. Acoust. Soc. Am. **134**, 176–184 (2013)]. The waves were generated by a low-frequency shear wave source on the seafloor, which operated in the 2–60 Hz band. Dispersion curves of the interface waves exhibit linear dependences between the logarithm of phase speed and logarithm of frequency, with distinct slopes at large and small phase speeds. This suggests a geoacoustic model consisting of two sediment layers with power-law dependences of the shear wave speed on depth below seafloor [O. A. Godin and D. M. F. Chapman, J. Acoust. Soc. Am. **110**, 1890–1907 (2001)]. In this paper, dispersion curves of interface waves are inverted for the shear wave speed profile in the sediments. Observations of lateral wave arrivals are used to corroborate the geoacoustic model suggested by the surface wave dispersion and constrain compressional wave speeds. Combining data for waves of different polarizations helps to identify various arrivals, check consistency of inversions, and evaluate sediment density.

TUESDAY AFTERNOON, 8 MAY 2018 GREENWAY F/G, 1:15 P.M. TO 5:00 P.M.

Session 2pBA

Biomedical Acoustics: Using Acoustic Wave Propagation to Estimate Quantitative Material Properties of Tissue II

Matthew W. Urban, Chair
Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905

Contributed Papers

1:15

2pBA1. Reconstructing one-dimensional positions of spherical scatterers from overlapping echoes. Aiguo Han (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 306 N. Wright St., Urbana, IL 61801, han51@uiuc.edu)

A new method for identifying the one-dimensional positions of spherical scatterers from overlapping echoes is described. This problem is of interest because various tissues (e.g., solid tumors) can be modeled as an ensemble of discrete scatterers (e.g., tumor cells). Existing quantitative ultrasound methods are successful in estimating the scatterer size by fitting an acoustic scattering model to the backscatter coefficient estimated from the radio-frequency data. However, the scatterer positions cannot be recovered from severely overlapped echoes using such quantitative ultrasound methods alone. In the method proposed herein, the forward scattering problem is formulated based on a sparse discrete scatterer assumption. The forward problem takes into account the effects of attenuation, scattering function, transducer diffraction and system response function, all of which can be estimated or corrected using quantitative ultrasound techniques. The scatterer positions are then recovered by solving the inverse problem using compressive sensing techniques. The detection limit (distances between scatterers and number of scatterers) is investigated through numerical simulations as a function of various real-world factors such as the signal-to-noise ratio. Super resolution capability is demonstrated. Physical phantom experiments performed using single-element transducers will be presented to evaluate the method. [Support: NIH R01DK106419 and NIH R37EB002641.]

1:30

2pBA2. Estimating liver fat fraction from ultrasonic attenuation and backscatter coefficient measures in adults with nonalcoholic fatty liver disease. Lucia Albelda, Aiguo Han (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), Ethan Z. Sy (Liver Imaging Group, Dept. of Radiology, Univ. of Califiornia at San Diego, La Jolla, CA), Michael P. Andre (Dept. of Radiology and the San Diego VA Healthcare System, Univ. of California at San Diego, La Jolla, CA), Rohit Loomba (NAFLD Res. Ctr., Div. of Gastroenterology, Dept. of Medicine, Univ. of California at San Diego, La Jolla, CA), Claude B. Sirlin (Liver Imaging Group, Dept. of Radiology, Univ. of Califiornia at San Diego, La Jolla, CA), John W. Erdman (Dept. of Food Sci. and Human Nutrition, Univ. of Illinois at Urbana-Champaign, Urbana, IL), and W. D. O'Brien, Jr. (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 306 N. Wright St., Urbana, IL 61801, wdo@uiuc.edu)

Between 80 and 100 million people in the United States suffer from nonalcoholic fatty liver disease (NAFLD). It is currently being diagnosed via MRI and biopsies; however, these tend to be expensive and painful. In this study, a Siemens 3000 ultrasound scanner was used to obtain the quantitative ultrasound (QUS) parameters [backscatter coefficient, BSC, and attenuation coefficient, AC] of 84 participants with known or suspected NAFLD (thus few uncompromised liver samples expected). Participant recruitment is continuing. Also obtained was the liver proton density fat fraction (PDFF) via MRI. AC and BSC individually were linearly correlated (~0.6) to PDFF. The data were then divided into two equally sized sets, training and test, in order to calculate PDFF from a function of AC and BSC. Using the training set a polynomial equation was created where AC and BSC were used to estimate PDFF. It was found that a polynomial 1,2 [AC,BSC] yielded an improved correlation (0.76) when evaluated using the test set. Significant improvement is obtained to estimate PDFF using both AC and BSC as opposed to using either of these QUS parameters separately. [Support: NIH R01DK106419 and NIH R37EB002641.]

1:45

2pBA3. Direct measurement of nuclear diameter with high-frequency ultrasound (10–100 MHz) for breast cancer detection. Timothy E. Doyle (Phys., Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999, Timothy.Doyle@uvu.edu), Jessica E. Carlson (Biology, Utah Valley Univ., Orem, UT), Nicole Cowan (BioTechnol., Utah Valley Univ., Orem, UT), and Garrett Wagner (Comput. Eng., Utah Valley Univ., Orem, UT)

Two high-frequency (10–100 MHz) ultrasound studies were conducted at the Huntsman Cancer Institute (Salt Lake City, Utah) on breast cancer margin and lymph node specimens from 90 patients, and produced sensitivities and specificities as high as 87.5% and 82.9%. The method used through-transmission measurements and a new parameter that correlated not only to malignant versus benign tissue, but also to various breast cancer types and other pathologies. This parameter, peak density, measures the number of peaks in the ultrasonic spectrum. The data indicated that peak density increased successively for atypical pathologies, ductal carcinomas, and lobular carcinomas as compared to normal tissue. Additional correlations were found between the distribution of peaks in the spectra and the above pathologies. This study's objective was to test the hypothesis that peak density measures the nuclear diameter of the cells in the tested tissue. A computer model of Mie scattering from a single, nucleated cell was used to calculate ultrasonic spectral peaks in the 10-100 MHz band as a function of nuclear diameter. Experiments were also conducted with agarose tissue phantoms containing polyethylene microspheres having diameters of 10-102 μm. The model and phantom results confirmed the hypothesis that peak density directly correlates to nuclear diameter.

2:00

2pBA4. Inter-sonographer reproducibility of ultrasonic attenuation backscatter coefficient measures in adults with nonalcoholic fatty liver disease. Aiguo Han (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 502 W Main St. Apt 129, Urbana, IL 61801, han51@uiuc.edu), Ethan Z. Sy (Liver Imaging Group, Dept. of Radiology, Univ. of California at San Diego, La Jolla, CA), Michael P. Andre (Dept. of Radiology and the San Diego VA Healthcare System, Univ. of California at San Diego, La Jolla, CA), Rohit Loomba (NAFLD Res. Ctr., Div. of Gastroenterology, Dept. of Medicine, Univ. of California at San Diego, La Jolla, CA), Claude B. Sirlin (Liver Imaging Group, Dept. of Radiology, Univ. of California at San Diego, La Jolla, CA), John W. Erdman (Dept. of Food Sci. and Human Nutrition, Univ. of Illinois at Urbana-Champaign, Urbana, IL), and W. D. O'Brien, Jr. (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

The attenuation coefficient (AC) and backscatter coefficient (BSC) measures in the liver were found repeatable and inter-transducer reproducible in previous studies. This study assesses the inter-sonographer reproducibility of the two parameters. 56 participants (sex: 20M, 36F; age: 51.5 ± 13.5, 24–74 yo; BMI: 32.2 ± 5.2, 23.9–47.4 kg/m^2; 43 had MRI-PDFF: 15.6 ± 10.4%, 0.7–41.1%) with known or suspected NAFLD each underwent two same-day ultrasound liver examinations performed by two sonographers (out of six in total) using the Siemens S3000® ultrasound system. Each examination included five RF acquisitions in separate breathholds from the right liver lobe and one from a calibrated phantom using the same settings. The AC and BSC were estimated using the reference phantom technique from each liver RF acquisition. AC and log-transformed BSC (10logBSC) are frequency-averaged within 2.6-3.0 MHz. Intraclass correlation coefficient (ICC) was used to evaluate the inter-sonographer reproducibility. Without averaging multiple intra-examination acquisitions, ICC (95% confidence interval) was 0.79 (0.65, 0.88) for AC, and 0.85 (0.74, 0.91) for 10logBSC. Averaging five intra-examination acquisitions increased ICC to 0.86 (0.77, 0.92) for AC, and 0.89 (0.81, 0.94) for 10logBSC. Theseresults show excellent inter-sonographer reproducibility of AC and BSC in patients with known/suspected NAFLD. [Support: R01DK106419 and Siemens Medical Systems.]

2:15

2pBA5. Vibro-acoustic method for detection of osteopenia and osteoporosis. Siavash Ghavami (Radiology, Mayo Clinic, College of Medicine, 200 First St. SW, Rochester, MN 55905, roudsari.seyed@mayo.edu), Mahdi Bayat, Adriana Gregory (Physiol. and Biomedical Eng., Mayo Clinic, College of Medicine, Rochester, MN), Jeremy Webb (Radiology, Mayo Clinic, College of Medicine, Rochester, MN), Max Denis (Physiol. and Biomedical Eng., Mayo Clinic, College of Medicine, Lowell, MA), Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, College of Medicine, Rochester, MN), and Azra Alizad (Radiology, Mayo Clinic, College of Medicine, Rochester, MN)

Early detection of osteopenia and osteoporosis is critical in identifying at-risk groups and manage necessary therapy. In this paper, a new noninvasive and quantitative vibro-acoustic method is proposed for detection of osteoporosis and osteopenia. In this method, we excite the bone by an ultrasound radiation force (URF) pulse. The URF pulse induces a vibration in the bone, resulting an acoustic wave that is received by a hydrophone placed on the skin. This prospective study included $n=23$ volunteers. The number of volunteers with osteopenia and osteoporosis were 7, 6, respectively, and the number of volunteers with normal bone was 10. URF excitations were applied at four equi-distance points on the bone. The resulting acoustic signals were used for wave velocity estimation based on cross-correlation technique. In general, the hydrophone signal may be modeled as a superposition of different modes of bone vibration. Based on this model, we used variational mode decomposition (VMD) as an efficient technique to decompose the received signal into an ensemble of band-limited intrinsic mode functions. Statistical analysis demonstrates that osteopenia and osteoporosis bones can be differentiated from normal bone with $p< 0.001$ using Wilcoxon rank-sum test on estimated velocity of the modes with maximum power.

2:30

2pBA6. Comparison of conventional transducer and refracto-vibrometry measurements of ultrasonic transmission parameters from heel bones. Thomas M. Huber, Matthew R. Mehrkens, Benjamin A. Rorem (Phys., Gustavus Adolphus College, 800 W College Ave., Saint Peter, MN 56082, huber@gac.edu), Matthew T. Huber, and Brent Hoffmeister (Phys., Rhodes College, Memphis, TN)

Ultrasonic measurements of the heel bone (calcaneus) are used commonly for osteoporosis screening. Ultrasound pulses that pass through the calcaneus are detected with a receiving transducer. Parameters, such as normalized broadband ultrasound attenuation (nBUA) and speed of sound (SOS) calculated from the wave forms detected by the receiving transducer, are utilized to assess bone health. In the current study, refracto-vibrometry (RV), an interferometric method for optically measuring ultrasound, was compared with conventional transducer measurements of ultrasound transmission *in-vitro* through a human calcaneus sample. The measurement beam from a Polytec PSV-400 scanning laser Doppler vibrometer was directed through a water tank towards a stationary retroreflective surface. Acoustic wave fronts (density variations) which pass through the ~50μm diameter measurement laser cause variations in the integrated optical path length. The signals detected by the vibrometer at numerous scan points were used to determine the time evolution of ultrasonic wave fronts. Because there is not a physical receiving transducer, the RV measurements do not experience the same frequency response and reflection artifacts of conventional transducers. Parameters such as nBUA and SOS were calculated at multiple RV scan points, and the results were compared to parameters measured using a conventional ultrasound transducer.

2:45

2pBA7. A continuous power law wave equation for biomedical ultrasound. James F. Kelly, Mark M. Meerschaert (Dept. of Statistics and Probability, Michigan State Univ., 1. East Lansing, MI 48824, kellyja8@stt.msu.edu), and Robert McGough (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

The attenuation coefficient in most biological media is a power law with respect to frequency. Measurements indicate that the power law exponent y is near or exactly one for many classes of tissue. In J. Acous. Soc. Am. [**124** 2861–2872 (2008)], a power law wave equation (PLWE) was proposed to model this power law dependence using Riemann-Liouville fractional derivatives. The PLWE, like most other fractional calculus models for attenuation in the time-domain, is invalid for y =1 due to a discontinuity in the phase velocity. To address this problem, a continuous power law wave equation (CPLWE) is proposed that is valid for all power law exponents between zero and two. The CPLWE utilizes the recently developed Zolotarev fractional derivative of order y, which is a nonlocal operator even for $y = 1$. The 3D Green's function for the CPLWE that is valid for homogeneous media is derived using stable probability density functions in the Zolotarev M-parameterization. Solutions to the CPLWE that are valid for inhomogeneous media are then expressed as solutions to the wave equation subordinated to an inverse stable density. These subordinated Green's function solutions transition smoothly between $y<1$ and $y>1$ power laws, which facilitates estimation of attenuation parameters in biological media with y close to one.

3:00–3:15 Break

3:15

2pBA8. Power law attenuation of shear waves in fractal media. Sverre Holm (Dept. of Informatics, Univ. of Oslo, Gaustadalleen 23B, Oslo N 0316 Oslo, Norway, sverre@ifi.uio.no) and Ralph Sinkus (Div. of Imaging Sci. and Biomedical Eng., Kings College London, London, United Kingdom)

There are two mechanisms for attenuation of mechanical waves: absorption and multiple scattering. We explore the second mechanism for shear waves in the context of MR elastography. The theory for attenuation was first given in the seismic field (O'Doherty & Anstey, Geophys. Prosp. 1971). Later phase was included, and the mean field concept was applied (Banik *et al.*, Geophys. 1985). Then, a more rigorous theory which also deals with random fractals was developed (Sølna, SIAM J ApplMath, 2003; Garnier & Sølna, Multiscale Model Simul, 2009). We have developed this theory further by allowing fractality over a limited length scale and also shown that it gives results that are indistinguishable from those of power law absorption models like the fractional Kelvin-Voigt model (Holm & Sinkus, JASA, 2010). This makes it a challenge to understand whether attenuation in a real medium is due to one or the other mechanism or both. Measurement of dispersion has been done for polystyrene microspheres embedded in agarose which is nearly lossless up to 1000 Hz. The phase velocity has a power law variation with exponent up to d = 0.24 (Lambert *et al.*, PRL 2015). According to the theory this is consistent with a fractional dimension which is d less than the Euclidean dimension.

3:30

2pBA9. Subharmonic generation by shear waves in a 1D resonator formed with a relaxing material. John M. Cormack and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

The low shear moduli of soft elastic media such as rubber and tissue facilitate the excitation of shear waves that exhibit significant finite-amplitude effects. Measurements performed by Andreev *et al.* [Acoust. Phys. **57**, 779 (2011)] of the amplitude-dependent response of a 1D shear-wave resonator formed with a rubber-like material revealed reasonable agreement with numerical simulations based on a monorelaxing material model. At the previous ASA meeting the present authors developed an augmented Duffing equation as a model for plane shear waves in a resonator formed with a monorelaxing material that is shaken at one end and free at the other [J. Acoust. Soc. Am. **142**, 2723 (2017)]. The focus of the previous work was on the response at drive frequencies near the lowest resonance. Here the augmented Duffing model is used to investigate subharmonic generation associated with a drive frequency of approximately three times that of the lowest mode. Conditions on the amplitude and frequency of excitation for which subharmonic generation can occur are obtained from the augmented Duffing equation. Results obtained from the augmented Duffing equation are compared with direct numerical solutions of a nonlinear wave equation. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

3:45

2pBA10. Parametric phantom study of shear wave velocity image reconstruction. Matthew W. Urban (Dept. of Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu) and Jorge Racedo (Dept. of Biomedical Eng. and Dept. of Phys., Universidad de los Andes, Bogota, Colombia)

Shear wave elastography is being increasingly used for multiple applications including investigation of diffuse disease in the liver or kidney and malignancies or neoplasms in the breast and thyroid. It is important to provide robust and accurate reconstructions of shear wave velocity (SWV) for imaging purposes to provide reliable data for diagnosis. One such approach was a two-dimensional method for estimating SWV (Song, *et al.*, Ultrasound Med. Biol. 40, pp. 1343–1355, 2014). This method uses a weighted average using a large window and smaller patches to perform time-domain cross-correlations for the calculation of the SWV. We also implemented a spatial sigmoid weighting to combine reconstruction results from independent reconstructions after directional filtering. We used a Verasonics system equipped with a linear array transducer to conduct a parametric study in phantoms to investigate the effects of the window and patch on image reconstruction metrics. Measurements were made in homogeneous phantoms and in a phantom with cylindrical inclusions of different diameters. Larger windows and patches typically yielded low variability but did not always produce the highest contrast. This investigation is the basis for an adaptive method for reconstructing SWV in soft tissues and choosing optimal parameters for the SWV reconstruction.

4:00

2pBA11. Quantifying nonlinear elasticity modulus of tissue-like solids using acoustic radiation force. Danial Panahandeh-Shahraki, Bojan B. Guzina (Dept. of Civil, Environ. and Geo- Eng., Univ. of Minnesota, 500 Pillsbury Dr. SE, Minneapolis, MN 55414, panah004@umn.edu), Viksit Kumar (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN), Matthew W. Urban (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., Rochester, MN), Randall R. Kinnick (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN), Siavash Ghavami, Azra Alizad (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., Rochester, MN), and Mostafa Fatemi (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN)

As shown in a recent study, estimation of the magnitude of the acoustic radiation force (ARF) in isotropic tissue-like solids is the key to measuring a third-order nonlinear modulus of elasticity (C)—responsible for coupling the shear and volumetric constitutive responses of a material. Due to experimental limitations, a direct measurement of the ARF in tissue is however difficult. To overcome this problem, we deploy 3D elastodynamic finite element (EFE) simulations of the ARF experiment in tissue-mimicking phantoms to estimate (i) the "background" shear modulus from the phase of the induced shear waves; (ii) the ARF magnitude from the shear wave amplitude, and (iii) the modulus C from the knowledge of the ARF. In this way, by interpreting the ARF-generated shear waves through the prism of 3D-EFE simulations, we facilitate local estimation of the nonlinear C modulus with spatial resolution equaling the size of the focal region. We call this method the C-Elastography (CE). In this study, the CE technique is applied to gelatin phantoms containing an agar-based inclusion. The profiles of the C-modulus in phantoms tested demonstrate good correlation with the phantom geometry, showing marked (and sharp) C-contrast at push points acting on the inclusion. [Acknowledgment: NIH-Grant EB23113.]

4:15

2pBA12. Finite element analysis of lung ultrasound surface wave elastography. Boran Zhou and Xiaoming Zhang (Radiology, Mayo Clinic, 321 3rd Ave. sw, Rochester, MN 55902, Zhou.Boran@mayo.edu)

Interstitial lung disease manifested as fibrotic and stiffened lung tissue, resulting in symptoms such as dyspnea and may lead to respiratory failure. We recently developed a lung ultrasound surface wave elastography for evaluating the elastic properties of superficial lung tissue. In this abstract, we present a finite element analysis to investigate the effects of pleural effusion on the shear wave propagation of superficial lung tissue. The muscle, pleural effusion, and lung tissue were simulated by a 2D planar model of an infinite elastic medium with densities for different materials. A nearly incompressible, linear elastic model was used to model muscle, pleural effusion and lung. The model was excited using a line source on the muscle surface. Harmonic excitations were performed at 100, 150 and 200 Hz with duration of 0.1 s. The boundaries of lung and muscle were attached to an infinite. The mesh of muscle, pleural effusion, and lung were constructed using linear quadrilateral elements. The infinite region was meshed by infinite elements (type CINPE4). The dynamic responses of the tissue model to the excitations were solved by the ABAQUS explicit dynamic solver with automatic step size control. Insignificant difference in shear wave speeds of lung tissue at different excitation frequencies was found in both simulations with and without taking into account the pleural effusion, indicating that pleural effusion does not affect the measurements of shear wave speeds in ultrasound elastography.

4:30

2pBA13. Pleural fluid may pose no measurable effect on lung stiffness in lung ultrasound surface wave elastography. Jinling Zhou and Xiaoming Zhang (Dept. of Radiology, Mayo Clinic, 1-24 Medical Sci. Bldg., 321 3rd Ave. SW, Rochester, MN 55902, zhou.jinling@mayo.edu)

Lung ultrasound surface wave elastography (LUSWE) was developed as a non-invasive tool for assessing pulmonary fibrosis. In this research, the effect of the fleural fluid on surface wave speed measurement of the lung is studied. Low-cost sponges [J. Ultrasound Med. 2017; 36, 2133], as good lung phantoms, were used in this study. The ultrasound transmission gel was used to simulate the pleural fluid. A harmonic vibration was generated at sponge surfaces using a shaker at 100, 150, and 200 Hz. The sponge surface wave speeds were measured with a 6 MHz ultrasound probe for both with and without a thin layer gel. The ultrasound features of A and B-lines were clearly observed in every sponge model in our experiment. The surface wave speeds for the sponges with the gel were indistinguishable from those without the gel for each frequency. The surface wave speeds were faster at higher excitation frequencies which is consistent with our patient study results. It was conclude that the pleural fluid may not affect the lung surface wave speed measurement based upon our lung phantom sponge models.

4:45

2pBA14. A deep neural network method for analyzing lung density using lung ultrasound surface wave elastography. Boran Zhou and Xiaoming Zhang (Radiology, Mayo Clinic, 321 3rd Ave. sw, Rochester, MN 55902, Zhou.Boran@mayo.edu)

Lung density is directly associated with lung pathology. Computed Tomography (CT) evaluates lung pathology using the Hounsfield unit (HU) but not directly the lung density. We recently developed a lung ultrasound surface wave elastography (LUSWE) technique for measuring the surface wave speed of superficial lung tissue. The objective of this study is to analyze lung density of superficial lung tissue using deep neural network (DNN) and wave speed measurements obtained from LUSWE. The synthetic training dataset (7,88,000 in total) in terms of wave speeds of superficial lung tissue at different excitation frequencies from LUSWE, viscoelasticity and density of lung tissue was used to train the DNN. DNN is composed of 3 hidden layers of 1024 neurons and trained for 10 epochs with a batch size of 4096 and a learning rate of 0.001 with Adam optimizer. Test dataset in terms of wave speeds at different excitation frequencies (100, 150, and 200 Hz) as well as elasticity was used to predict the lung density and evaluate its accuracy. The obtained results showed that predictions matched well with test dataset (validation accuracy is 0.997). This method may be useful to analyze lung density based on the LUSWE measurements.

TUESDAY AFTERNOON, 8 MAY 2018 GREENWAY D, 1:00 P.M. TO 2:15 P.M.

Session 2pEA

Engineering Acoustics and Physical Acoustics: Miniature Acoustic Transducers II

Vahid Naderyan, Chair
Physics/National Center for Physical Acoustics, University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677

Contributed Papers

1:00

2pEA1. Artificial design of open-back headphone based on ventilation structure optimization method. Yu-Ting Tsai (Feng Chia Univ., No. 100 Wenhwa Rd., Seatwen, Taichung City 40724, Taiwan, yuttsai@fcu.edu.tw) and Jin H. Huang (Feng Chia Univ., Taichung City, Taiwan)

The computer-aided analysis of open-back headphone based on finite/bound element method, lumped parameter model, and transduction equations have been investigated for decades. In a serious attempt to reach the optimal design of headphone, various aspects of conceptualization, design, modeling, and simulation are investigated for improvement of headphone in this work. The standardized construction process of the three-dimensional image of the human ear to establish the open-back architecture for the headphone is given. For improving the sound pressure level response of the headphone, the parameter design method is proposed through the optimized dimension modification of special open-cavity structure. All results of studies of the simulated responses are compared with experimental data and have verified the veracity of the simulations. The present work has concluded the feasibility of adjusting the model and parameters for customizing the resulting response for headphone's varied applications.

1:15

2pEA2. Frequency response simulation of the hybrid dual driver earphone using equivalent circuit model. Yu-Ting Tsai (Feng Chia Univ., No. 100 Wenhwa Rd., Seatwen, Taichung City 40724, Taiwan, yuttsai@fcu.edu.tw)

By combining the multiple drivers into an insert earphone, the special filtering of hybrid dual drivers by earphone structure design can be applied to increase efficiency and the overall output level. In general, the back of the earphone structure lays a balanced armature driver for the mid to high frequency reproduction and a dynamic driver in front cavity creates deeper bass. For this distinctive structure of the earphone, this study proposed a frequency response simulation model using the equivalent circuit model. For the sound transmission between the two ports' structure, the correction factors for obtaining the transmitted impedance of special filter are proposed. The simulated responses shows that a good comparison between the measured result and the equivalent circuit simulation. As a result, the study has significant capacity to estimate the desired frequency response of a hybrid dual driver earphone.

1:30

2pEA3. Miniature implantable low noise piezoelectric diaphragm sound sensor. Chuming Zhao, Alison Hake (Mech. Eng., Univ. of Michigan, 2350 Hayward, Ann Arbor, MI 48109, chumingz@umich.edu), Wang-Kyung Sung (Vesper Technologies Inc., Boston, MA), and Karl Grosh (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Piezoelectric diaphragm sound sensors must be miniaturized and environmentally toughened for use in biomedical applications. We study two disparate applications of the sensor, first as the front end of a completely implantable cochlear implant and the second as an intracardiac pressure sensor. The sensor, which is fabricated using micro-electrical-mechanical-system (MEMS) techniques, consists of a circular diaphragm with at least one aluminum nitride (AlN) based piezoelectric layer backed by an air-filled cavity. The sensors are designed to have a lower input referred noise (IRN) compared to other miniaturized piezoelectric diaphragm hydrophones with similar diaphragm size via structural design and electrode area optimization. Sensors are fabricated with varying diaphragm diameters from 300 to 450 μm to test the effect of residual stresses due to the manufacturing process on the frequency bandwidth and IRN. The sensor works in air and underwater with the same sensitivity, but the frequency bandwidth is reduced underwater compared to that is in air because of the water mass loading. A finite element analysis model using COMSOL is built to study the in air and underwater sensing process and the IRN.

1:45

2pEA4. Experimental assessment of miniature accelerometers designed for sensing middle ear motion. Alison Hake, Chuming Zhao (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, aehake@umich.edu), Wang-Kyung Sung (Vesper Technologies, Medford, MA), and Karl Grosh (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Sensing human middle ear motion requires miniature, low-noise sensors to detect low-amplitude vibration of the ossicle bones to replace the external microphone in a cochlear implant system. Specifically, a 35 $\mu g/\sqrt{Hz}$ sensor resolution is needed measure motion at 500 Hz (Young *et al.*, 2012). Motivated by sensing middle ear motion without irreversible alteration to the ossicular chain, a small, piezoelectric accelerometer is considered. Our detailed low-frequency analytical model is used to identify promising micromachined, piezoelectric bimorph cantilever accelerometer designs. We fabricated a first generation of accelerometers using a conservative microfabrication approach, taken to enhance the yield of devices rather than to minimize the input referred noise (IRN). The mathematical model was validated by voltage actuation testing of these accelerometers. The device sensitivity to acceleration is tested, and the resulting IRN is desired to assess the viability of the sensor for the middle ear motion application. The first generation experimental results are compared to the mathematical model to determine the accuracy of the model and the efficacy of the fabrication methodology. This knowledge will influence future sensor designs optimized for minimal IRN.

2:00

2pEA5. A new approach for an FE-model optimization of a MEMS piezoelectric accelerometer for implantable hearing devices. Zargos Neves Masson, Andre L. Gesing, Matheus de Lorenzo (Mech. Eng., Federal Univ. of Santa Catarina, Av. Madre Benvenuta,40, ap 302, Florianopolis, Santa Catarina 88036500, Brazil, zargosm@gmail.com), Stephan Paul (Mech. Eng., Federal Univ. of Santa Catarina, Joinville, Brazil), and Julio A. Cordioli (Mech. Eng., Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil)

Totally implantable hearing devices have been proposed as a solution to mitigate the constraints related to the presence of external elements in traditional hearing devices. This work reports on a novel approach for the development of a MEMS piezoelectric accelerometer as an implantable sensor for hearing devices. Traditionally these sensors are designed for maximum sensitivity; however, the primary bottleneck for this type of transducer is the internal noise level. Two differential evolution optimization routines were developed. The first used sensitivity maximization whereas the second seeks minimization of the equivalent input noise (EIN). Both methods were applied to different designs of Lead Zirconate Titanate (PZT) MEMS piezoelectric accelerometer in the frequency range of 250 Hz to 8 kHz with layer thickness varying from 0.1 μm to 1 μm. In the latter approach, the sensor's acceleration noise was estimated analytically, considering capacitance and charge response acquired through finite element modeling (FEM) previously validated. Acceleration noise was converted to EIN, in sound pressure level (SPL), through an FE-model of the middle ear considering the sensor coupled at the umbo. Preliminary results indicate that using EIN as optimization goal, opposed to sensitivity, leads to a higher performance over a broader bandwidth.

Session 2pNS

Noise and ASA Committee on Standards: Effects of Natural Soundscapes on Recreation Areas

David Braslau, Cochair
David Braslau Associates, Inc., 6603 Queen Ave. S, Suite N, Richfield, MN, MN 55423

Kurt M. Fristrup, Cochair
Natural Sounds and Night Skies Division, National Park Service, 1201 Oakridge Drive, Suite 100, Fort Collins, CO 80525

Chair's Introduction—1:00

Invited Papers

1:05

2pNS1. Conflicting sounds in natural recreation areas—An historical Minnesota perspective. David Braslau (David Braslau Assoc., Inc., 6603 Queen Ave. S, Ste. N, Richfield, MN 55423, david@braslau.com)

Minnesota has long seen battles between conflicting activities in natural recreation areas. Snowmobiles were first introduced in the 1960 by two Minnesota firms. By the 1970s physical and noise impacts had increased and a state committee to address trail uses and conflicts was established. In the same time period, a large scale environmental impact study on effects of copper-nickel mining on the BWCAW and national forest was initiated. A noise model was developed by Moorhead State staff that compared mining noise with natural noise levels. This issue rested until 2007 when exploration drilling started 24/7 in the Superior Nation Forest and near the BWCAW. Extensive monitoring and modeling efforts were undertaken by the companies and the Forest Service and limits established on acceptable noise levels. The snowmobile noise problem has not gone away with periodic controversy and assessment. To the south, the Saint Croix National Scenic Riverway (NPS) and the Minnesota Valley National Wildlife Refuge (USFWS) have both addressed potential noise intrusion from adjacent sand and gravel mining activities. Of course, the BWCAW was the center of long term disputes over noise, with motors now limited to only a few lakes and low altitude aircraft flights prohibited.

1:25

2pNS2. Management considerations for noise impacts on the superior National Forest and Boundary Waters Canoe Area Wilderness. Peter Taylor and Ann Schwaller (Superior National Forest, Forest Headquarters, 8901 Grand Ave. Pl., Duluth, MN 55808, prtaylor@fs.fed.us)

The USDA Forest Service-Superior National Forest manages the Boundary Waters Canoe Area Wilderness (BWCAW) and multiple use lands located outside the Wilderness. For the BWCAW, the Forest Service is responsible for preserving wilderness character per Section 4b of the 1964 Wilderness Act. Accordingly, Forest Service managers need to understand the effects of human-generated sound on the soundscapes of the Wilderness when making decisions on if, where, and how to permit management activities to occur. Human-generated sound that may affect the Wilderness has been a consideration in management decisions for timber harvest, snowmobiling, ATVing, minerals exploration, and mining. Further, human-generated sound may also be an issue for recreation users or residents located on multiple use lands outside the Wilderness. We discuss the ongoing need for scientific information to inform decision making on this issue. We also discuss our monitoring efforts and data and analysis needs.

1:45

2pNS3. Modeling and mapping aircraft noise in National Parks. Amanda Rapoza (Volpe National Transportation Systems Ctr., U.S. Dept. of Transportation, 55 Broadway, RVT-41, Cambridge, MA 02142, amanda.rapoza@dot.gov), Brent Lignell, Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO), and Aaron L. Hastings (Volpe National Transportation Systems Ctr., U.S. Dept. of Transportation, Cambridge, MA)

Commercial air tours are a common source of noise within many national parks, affecting natural and cultural resources as well as visitor experience. To facilitate interactive modeling and mapping of air tour noise exposure, the Volpe National Transportation Systems Center and National Park Service (NPS), in cooperation with the Federal Aviation Administration (FAA), have created an Aviation Noise Analysis and Mapping Tool. This tool contains a library of park noise maps generated by modeling single aircraft operations on routes flown by that aircraft. The creation of this library is computationally intensive and requires considerable expertise to properly configure and run the models. With this tool and library, a broad range of users can interactively explore composite noise exposure generated by air tour traffic scenarios. Composite noise exposure is calculated by summing, across all aircraft and route combinations, the product of the number of flights on each route and the noise generated by each aircraft on that route. The use of FAA noise models ensures consistency with other U.S. environmental analyses, use of state-of-the-art algorithms, and access to updated aircraft noise databases. The traffic aggregation component can be modified to incorporate specialized interpretations of noise exposure for NPS applications.

2:05

2pNS4. Forecasting increases in recreational value that would result from restoration of natural soundscapes in National Parks. Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CA 80525, kurt_fristrup@nps.gov), Megan F. McKenna, and Daniel J. Mennitt (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

U. S. National Parks are justly celebrated for the superlative quality of their scenic and cultural resources. The Grand Canyon Enlargement Act of 1975 identified natural quiet as an important resource and value. These considerations are two motivations for extensive monitoring of acoustic environments in national park units, and development of models that predict sound levels throughout the system. These models also estimate the spatial distribution of noise of noise exposure. Related social science research has documented the effects of noise on the quality of visitor experience in parks, including the accessibility of wildlife for public viewing. Surveys of park visitors and the American public also document broad support of park efforts to reduce noise. Collectively, this research provides a quantitative framework for estimating the increases in the quality of visitor experience that could be realized through reductions in noise.

2:25–2:40 Break

Contributed Papers

2:40

2pNS5. "Listening" Crowdsourced Sound. Yalcin Yildirim (Urban Planning and Public Policy, Univ. of Texas at Arlington, 601 W Nedderman Dr #203, Apt. 933, Arlington, TX 76019, yalcin.yildirim@mavs.uta.edu)

Bearing in mind the influence of internet based instruments for crowdsourcing, landscape architecture, urban planning, and urban design fields are progressively applying these tools to obtain better notions and alternatives from the community. Such instruments generally provide considerable data about what community desires. In spite of this fact, to the best of our knowledge, crowdsourced intelligence in landscape architecture and urban planning has been studied limited. This research concentrates on University of Texas at Arlington campus soundscape in the heart of 6 million populated Dallas-Fort Worth metroplex to examine the opportunities and applications of performing crowdsourced data. To do this, research team applied mixed methods, the study evaluates the opinions of campus users and determines in which aspects those ideas can be connected to soundscape patterns. After investigating the interviews of campus users, the study integrated the information of perceptions about opinions at the end of the research. The findings emphasizes the challenges, limitations, and opportunities in regard to the landscape architecture, urban planning, and urban design disciplines. It is noteworthy that several circumstances have implications on applicability of crowdsourced knowledge on soundscape.

2:55

2pNS6. A basic research on expression of loudness and orientation of sound source in soundscape by onomatopoeia on pictures. Takeshi Akita (Dept. of Architecture, School of Sci. and Technol. for Future Life, Tokyo Denki Univ., 5 Senju-Asahi-cho Adachi-ku, Tokyo 1208551, Japan, akita@cck.dendai.ac.jp), Takaaki Koga (Utsunomiya Univ., Utsunomiya, Japan), Naoko Sano, and Ayako Matsuo (Tokyo Denki Univ., Adachi-ku, Tokyo, Japan)

When soundscape is described with words under common acoustic environment, it is usual that the name of the sound source that indicates their category or species is noted. Generally, such kind of description cannot express loudness and orientation of sound source. However, the information of loudness and orientation of the sound source is important, so it is necessary to note them easily at a time. In the present research, it is investigated whether description of soundscape using onomatopoeia on pictures to express loudness and orientation of sound sources is effective or not. Two psychological experiments are carried out to reveal the availability of this description method. Results show that onomatopoeia can present the type of source, and that it can simultaneously express variety of loudness of sound source and noise. It is suggested that the size and number of onomatopoeia described on pictures of the place where noise exists represents loudness of sound and noise, and that the variety of size of onomatopoeia can show the difference of sound pressure level of 20 dB.

3:10

2pNS7. Social investigation on passersby perception of urban fountain sounds. Laura Velardi (LISA - Environ. HydroAcoust. Lab, ULB- Université libre de Bruxelles, av. F.D. Roosevelt 50, Brussels 1050, Belgium, lvelardi@ulb.ac.be) and Jean-Pierre Hermand (LISA - Environ. HydroAcoust. Lab, ULB- Université libre de Bruxelles, Brussels, Brussels Capital, Belgium)

Running water sounds are among the most pleasant sounds to human ears. Beside natural environments, people experience these sounds in urban environments thanks to fountains. The influence of fountain sounds on passersby has little been explored, especially *in situ*. This paper presents results from social surveys conducted in Rome and Brussels' squares for monumental and contemplative fountains as well as modern and participative fountains. Over 70 % of survey respondents qualified the fountain sounds as the most interesting feature of the place, even though the questionnaire did not make any explicit reference to acoustics. For each case study, a comprehensive set of timbre descriptors was applied to audio recordings in the fountain surroundings to characterize sound pleasantness in relation to people's impressions. Beyond their marked interest for the fountain soundmark, the survey identified people's preferences in terms of arrangement of fountain surroundings. Such information can be useful to forthcoming urban acoustic design, currently lacking any in depth observation about pedestrian land use related to urban sounds.

3:25

2pNS8. Noise dependent coherence-super Gaussian based dual microphone speech enhancement for hearing aid application using smartphone. Nikhil Shankar, Gautam Shreedhar Bhat, Chandan Karadagur Ananda Reddy, and Issa M. Panahi (Elec. and Comput. Eng., Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, nxs162330@utdallas.edu)

In this work, the coherence between speech and noise signals is used to obtain a Speech Enhancement (SE) gain function, in combination with a Super-Gaussian Joint Maximum a Posteriori (SGJMAP) single microphone SE gain function. The proposed SE method is implemented on a smartphone that works as an assistive device to hearing aids in real-time. Although coherence SE gain function suppresses the background noise well, it distorts the speech. In contrary, SE using SGJMAP improves speech quality with the introduction of musical noise, which we contain by using a post filter.

The weighted union of these two gain functions strikes a balance between noise suppression and speech distortion. A "weighting" parameter is introduced in the derived gain function that can be controlled by the smartphone user based on different background noises and their comfort level of hearing, which proves the practical usability of the developed SE method implemented on the smartphone. Objective and subjective measures of the proposed method show significant improvements in both quality and intelligibility compared to the state of the art SE techniques considered in this work for several noisy conditions at the signal to noise ratio levels of -5 dB, 0 dB, and 5 dB.

TUESDAY AFTERNOON, 8 MAY 2018 GREENWAY J, 1:00 P.M. TO 4:10 P.M.

Session 2pPA

Physical Acoustics: Infrasound for Global Security II

Philip Blom, Chair
Los Alamos National Laboratory, Los Alamos National Laboratory, PO Box 1663, Los Alamos, NM 87545

Contributed Papers

1:00

2pPA1. Acoustic waveform inversion and uncertainty analysis for explosion yield estimation. Keehoon Kim and Arthur Rodgers (Geophysical Monitoring Program, Lawrence Livermore National Lab., 7000 East Ave., L-103, Livermore, CA 94550, kim84@llnl.gov)

Waveform inversion techniques are widely used to extract source information from acoustic signals. The methods exploit full-waveform information and provide further constraints on source inversion than other parameter-based inversions. Infrasound waveform inversion was recently applied to chemical explosions and showed promising results for the determination of explosion yield. The method can improve the accuracy of the estimated yield by using full 3-D numerical Green's functions and incorporating the propagation path effects into the inversion. However, the inversion results are often given without any uncertainty estimation, which is critical for a post-detonation forensic analysis. In this study, we evaluate the yield estimation errors associated with the uncertainty in the atmospheric representation. An ensemble of atmospheric realizations is obtained by adding random turbulence to an atmospheric model and the uncertainty of inversion solution depending on atmospheric variation is quantified. In addition, different atmospheric models derived from local weather data or numerical weather prediction models are validated for yield estimation by using a series of chemical explosion data.

1:15

2pPA2. Estimation of explosive yield using regional distance infrasound. Philip Blom, Fransiska K. Dannemann, and Omar Marcillo (Los Alamos National Lab., Los Alamos National Lab., PO Box 1663, Los Alamos, NM 87545, pblom@lanl.gov)

Infrasound signals from large above-ground explosions are known to propagate to significant distances while remaining at observable levels due to a combination of the large source energy for such events and decreased absorption for acoustic energy at lower frequencies. Previously, propagation-based, stochastic path geometry and travel time models have been used to improve estimates of location and time for infrasonic sources with promising results. A similar approach has been applied to develop stochastic transmission loss models for infrasonic signals. The resulting models have been utilized in a Bayesian framework to compute a near-source estimate of the acoustic signal, which can then be used to compute a probability distribution for possible explosive yields of the source. Details of the propagation models and yield estimation framework will be presented along with results for application to a number of explosive events.

1:30

2pPA3. Improvements to infrasonic detection capabilities through application of a generalized least squares beamformer. Fransiska K. Dannemann, Philip Blom, and Omar Marcillo (Earth and Environ. Sci., Los Alamos National Lab., P.O. Box 1663, MS D446, Los Alamos, NM 87545, fransiska@lanl.gov)

A common challenge in seismoacoustic research and analysis is detection of low signal-to-noise ratio (SNR) signals in a variable and often coherent background. Previously, the application of an adaptive F-detector to infrasonic data has successfully reduced false detection rates attributed to coherent noise across array elements; however, the detector is applied post-processing after analysis using a standard (Bartlett) beamformer, which raises the detection threshold and can lead to missed detections. Application of a generalized least squares (GLS) beamformer can enhance processing of transient infrasonic signals, particularly in the presence of correlated noise, by adaptively accounting for the background noise characteristics. The GLS beamformer enhancement leads to improved capabilities for accurately detecting low amplitude infrasonic events of interest that would not normally produce a sufficiently high F-statistic to be declared a detection. A statistical analysis of GLS beamformer performance in a variety of noise environment will be presented, along with the statistical significance of enhanced detection capabilities compared to traditional beamforming approaches.

1:45

2pPA4. Realistic terrain boundary conditions for the three-dimensional finite-difference parabolic equation method applied to infrasonic propagation. Codor Khodr, Mahdi Azarpeyvand (Mech. Eng., Univ. of Bristol, 66B Queen's Rd., Bristol, Avon BS8 1QU, United Kingdom, ck15491@bristol.ac.uk), and David Green (AWE Blacknest, Reading, United Kingdom)

Infrasound propagation in the atmosphere can be significantly influenced by environmental parameters such as meteorology and topography. An accurate modelling of these parameters and their effects can improve signal processing of infrasound arrivals and development of more accurate prediction tools. Over the last few decades, much attention has been given to the integration of complex atmospheric features, namely, absorption, turbulence, density, and refraction, into the numerical methods, while topographic effects have usually been neglected. In this respect, the Parabolic Equation (PE) method has been continuously improved to deal with complex configurations encountered in various applications, such as atmospheric

waveguides, ocean acoustics and electromagnetics. We have further extended the Beilis-Tappert (BTPE) generalized coordinate transform to three-dimensional propagation in order to account for out-of-plane scattering by irregular three-dimensional topography, in the presence of air density variation with altitude. The resulting PE method is then implemented into a numerical solver and used to model wave propagation over a set of generic and realistic terrain and atmospheric conditions. A comparison against two-dimensional PE simulations allows us to isolate three-dimensional effects enabled by the out-of-plane propagation and to better understand the underlying physics of atmospheric wave propagation with complex terrain and atmospheric conditions.

2:00

2pPA5. An acoustic source model for the signal produced by ground motion over an underground event in mountainous terrain. Roger M. Waxler and Claus Hetzer (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu)

A confined, large explosion underground launches an infrasonic signal into the atmosphere generated by the heaving of the ground over the explosion. For example, the last three North Korean nuclear weapon tests produced infrasonic signals detected at great distances form the test site. Under flat terrain, modeling the acoustic source from a confined explosion is a classical problem, similar to that of a baffled loudspeaker. In mountainous regions, the problem is more difficult. A boundary element method has been developed to this end. Given the vertical motion of the ground surface, the model predicts an the acoustic pressure on a spherical section centered over the source. From this, a source function for far field propagation can be extracted. The methods used to produce such a model will be presented.

2:15

2pPA6. Scaling of a gas-combustion infrasound source. Chad M. Smith and Thomas B. Gabrielson (Graduate Program in Acoust., Penn State Univ., PO Box 30, State College, PA 16804, tbg3@psu.edu)

For equal storage pressure, the available energy density from propane-air combustion is substantially higher than the available energy density from expansion of a compressed gas. Ordinary liquid propane is stored at a pressure of about 1 MPa (150 psi). The potential-energy density in air that has been compressed to 1 MPa is about 170 kJ/kg; the combustion-energy density in propane is about 50 000 kJ/kg. In addition, there is a substantial volumetric advantage to propane in that it condenses to a liquid under normal storage pressures. Given the successful demonstration of compressed-gas infrasound sources (Barlett *et al*., J. Acoust. Soc. Am. **141**, 3567, 2017), useful levels of infrasound might also be produced by cycling a burner on and off. Previous work (Smith and Gabrielson, J. Acoust. Soc. Am. **137**, 2407, 2015) with a 90 kW propane burner demonstrated the potential for generating infrasound by periodic thermal expansion of the surrounding air and the levels measured were supported by a simple thermodynamic model for the periodic-heating process. In this paper, we describe attempts to scale these results to higher levels by using a liquid-propane burner from a hot-air balloon.

2:30–2:45 Break

2:45

2pPA7. Characterizing ocean ambient noise using infrasound network. Alexis Le Pichon, Marine De Carlo (CEA, DAM, DIF, -, Arpajon F-91297, France, alexis.le-pichon@cea.fr), Fabrice Ardhuin (IFREMER, LPO/CNRS, Brest, France), Thibault Arnal (CEA, DAM, DIF, Arpajon, France), and Lars Cerana (BGR, B4.3, Hannover, Germany)

The ability of the International Monitoring System (IMS) global infrasound network to detect atmospheric explosions and events of interest strongly depends on station specific ambient noise signatures which include both incoherent wind noise and real coherent infrasonic waves. To characterize the coherent ambient noise, broadband array processing was performed on historical continuous IMS recordings. Ocean wave interactions contribute to the atmospheric coherent ambient noise field, and we apply wave action models to model these microbarom sources. We use two-dimensional energy spectrum ocean wave products to build a global reference database of oceanic noise sources. Then, we compare observed and modeled directional microbarom amplitudes at several stations. The expected benefits of such studies concern the use complementary data to finely characterize the coupling mechanisms at the ocean-atmosphere interface. In return, better knowledge of ambient ocean noise sources opens new perspectives not only by enhancing the characterization of explosive atmospheric events, but also by providing additional integrated constraints on middle atmosphere dynamics and disturbances where data coverage is otherwise sparse.

3:00

2pPA8. Monitoring infrasound from a Tornado in Oklahoma. Brian R. Elbing, Christopher Petrin (Mech. & Aerosp. Eng., Oklahoma State Univ., OSU-MAE, Eng. North 218, Stillwater, OK 74078, elbing@okstate.edu), and Matthew S. Van Den Broeke (Earth & Atmospheric Sci., Univ. of Nebraska, Lincoln, Lincoln, NE)

Tornado-producing storm systems emit infrasound (sound at frequencies below human hearing) up to 2 hours before tornadogenesis. Weak atmospheric attenuation at these frequencies allows them to be detected hundreds of miles away. Hence, passive infrasonic monitoring may be used for long-range study of tornadogenesis as well as characterization of tornado properties if the infrasound can be correlated with flow-field properties. This requires characterization of infrasound during the life of a tornado as well as other background sources. This is being accomplished as part of the Collaboration Leading Operational UAS Development for Meteorology and Atmospheric Physics (CLOUD-MAP) project, a multi-university collaboration focused on the development and implementation of unmanned aerial systems (UAS) and their integration with sensors for atmospheric measurement. The current work focuses on analysis of a small tornado that occurred on May 11, 2017, about 19 km from an infrasonic monitoring station at Oklahoma State University. Results from this tornado will be reported as well as the available radar data.

3:15

2pPA9. Wind noise reduction at infrasound frequencies using large domes. Carrick L. Talmadge (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38655, clt@olemiss.edu)

One ongoing challenge to the use of infrasound for long-range monitoring applications (e.g., remote explosions or natural sources of infrasound such as tornados, volcanos or hurricanes) continues to be the high noise floors experienced at many permanent infrasound stations. International Monitoring System (IMS) infrasound stations typically address this problem using large pipe arrays as wind noise filters. However, these arrays generate significant distortion of higher frequency signals, making them non-ideal signals with significant frequency content above a few Hz. Further, they do not produce sufficient noise reduction during windy (typically daytime) measurements, making their utility generally limited to very quiet wind-noise periods (typically nighttime). Here we report on the development history of a 20-foot dome intended for permanent infrasound installations, and on measurements collected from arrays at the University of Mississippi Biological Field Station and at the Sandia National Laboratory Facility for Acceptance, Calibration, and Testing (FACT) site. The 20-foot dome at the FACT site is collocated with an IMS 18-m rosette filter and a University of Alaska Fairbanks experimental array, and more than a year of data has been collected, allowing for detailed comparisons of the efficacy of these different wind-noise reduction schemes.

3:30

2pPA10. Determining essential scales and associated uncertainties for regional atmospheric infrasound propagation by incorporating three-dimensional weather model forecasts. Ross E. Alter (Cold Regions Res. and Eng. Lab., US Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, ross.e.alter@usace.army.mil), Michelle E. Swearingen (Construction Eng. Res. Lab., US Army Engineer Res. and Development Ctr., Champaign, IL), and D. K. Wilson (Cold Regions Res. and Eng. Lab., US Army Engineer Res. and Development Ctr., Hanover, NH)

Understanding infrasound propagation is important for geophysical and military applications. Infrasound signatures can be detected from larger

sources such as nuclear detonations and from smaller sources such as bridges, dams, and buildings. Infrastructure sources produce signals of lower amplitude, leading to more regional (up to 150 km) propagation. However, current methods for calculating regional infrasound propagation involve assumptions about the atmosphere, such as horizontal homogeneity, that deviate from more realistic environmental conditions and decrease the accuracy of the infrasound predictions. To remedy this issue, we have interfaced three-dimensional forecasts from the Weather Research and Forecasting (WRF) meteorological model with range-dependent parabolic equation propagation models. To test the improvement of infrasound propagation predictions with more realistic weather data, we conducted sensitivity studies with different propagation ranges and horizontal resolutions and compared them to predictions using simplified meteorological parameters. This allowed identification of the scales most ideal for resolving regional infrasound propagation given the limitations of WRF's spatiotemporal resolutions. Additionally, we present results on quantifying uncertainty in these infrasound simulations by using multiple realizations of WRF forecasts to generate a spread of possible outcomes. Finally, we compare the simulated results to experimental data. (Approved for public release; distribution is unlimited.)

3:45–4:10 Panel Discussion

TUESDAY AFTERNOON, 8 MAY 2018 NICOLLET D2, 1:00 P.M. TO 4:55 P.M.

Session 2pPPa

Psychological and Physiological Acoustics and Animal Bioacoustics: Honoring the Contributions of David Kemp to the Discovery of Otoacoustic Emissons and their Utility for Assessing Hearing Function

Glenis R. Long, Cochair
Speech-Language-Hearing Science, Graduate Center CUNY, Graduate Center CUNY, 365 Fifth Ave., New York, NY 10016

Bastian Epp, Cochair
Electrical Engineering, Centre for Hearing and Speech Sciences, Technical University of Denmark, Ørsteds Plads, Building 352, Room 118, Lyngby 2800, Denmark

Chair's Introduction—1:00

Invited Papers

1:05

2pPPa1. Active cochlear mechanics and outer hair cells. Jonathan F. Ashmore (UCL Ear Inst., UCL, UCL Ear Inst., 332 Gray's Inn Rd., London WC1X8EE, United Kingdom, j.ashmore@ucl.ac.uk)

David Kemp's discovery of otoacoustic emissions signalled a new era in cochlear modelling and neurobiology. After Bekesy, the emphasis on cochlear processing had centered on the physics of frequency selectivity with any underlying physiological or cellular contributions taking a back seat. Since then, many lines of evidence have revealed that the (at the time) enigmatic population of outer hair cells (OHCs) within the cochlea plays a central role in modifying the basilar membrane mechanics: OHCs act as fast actuators up to at least 80 kHz in order to cancel fluid damping of the partition. The active mechanical feedback due to OHCs appears to be delicately poised so that small variations from section to section provides a substrate for the inhomogeneities required for OAEs. Metabolically labile, the intrinsic non-linearities of the system can be traced to the hair cell mechano-electrical transduction step itself. Mammalian OHC "motor" function has an identified molecular basis, and only recently have structural studies partially clarified how it this molecule ("prestin"/SLC26A5) works. A further contemporary concern is whether the mammalian cochlea employs the same, but scaled, mechanism of sound amplification for high as well as for low frequencies and this issue will be discussed.

1:25

2pPPa2. Otoacoustic emissions: Laboratory studies and clinical applications. Stephen T. Neely, Daniel M. Rasetshwane, and Michael P. Gorga (Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, Stephen.Neely@boystown.org)

Historically, the clinical potential of otoacoustic emissions (OAEs) to diagnosis hearing status was recognized prior to any consensus regarding their cochlear origin. In retrospect, we now understand refection-generated emissions as being closely aligned with Kemp's (1980) concept of "cochlear echoes," while distortion-generated emissions are necessarily tied to nonlinearities in cochlear mechanics. Research at the Boys Town National Research Hospital provided evidence supporting the cochlear origin of OAEs by demonstrating an approximate two-times relationship between an estimate of forward-latency derived from auditory brainstem responses and the round-trip latency of OAEs for tone-burst stimuli. Our application of clinical decision theory to distortion-product OAE levels, based on a large

number of both normal-hearing and hearing-impaired ears, helped to establish (1) the validity of using DPOAEs to identify the presence of hearing loss and (2) guidelines for the interpretation of DPOAE levels in the clinic. Our exploration of reflection-generated emissions that we call cochlear reflectance is a continuation of our efforts to further extend the clinical potential of OAEs.

1:45

2pPPa3. The echo that rolled into thunder: Otoacoustic emissions and newborn hearing screening. Beth Prieve (Syracuse Univ., 805 S. Crouse Ave, Syracuse, NY 13244, baprieve@syr.edu)

Kemp's 1978 publication that described "stimulated acoustic emissions" changed the basic understanding of the cochlea and provided a window to assess cochlear mechanics non-invasively in humans. The emissions, referred to early on as "Kemp's echoes," were adopted as the measure of choice by the Rhode Island Hearing Assessment Screening Program, a clinical research study that boldly investigated the ability to screen every baby for hearing loss. This presentation will review how a new, slightly suspicious, physiological measure was used as an integral part of improving hearing healthcare in newborns. The use of distortion-product and transiently evoked otoacoustic emissions as screening tools to identify hearing loss in newborns and their use in the differential diagnosis of hearing loss will be discussed. Clinical findings from infants with and without hearing loss will be presented that support current otoacoustic emission models, and new measures tied to cochlear physiology will be suggested that could lead to improved clinical use. This presentation emphasizes that the science of basic measures from the auditory system has fueled the national and international movement of improving hearing healthcare and habilitation in newborns.

2:05

2pPPa4. David Kemp's contributions to using otoacoustic emissions for protection of noise-exposed ears. Lynne Marshall (U.Conn. & Naval Submarine Med. Res. Lab, 118 Pearl St., Noank, CT 06340-5733, NoankLynneW@gmail.com)

In 1978, David Kemp published a paper that demonstrated a new auditory phenomenon, OAEs. Since then, OAE research expanded broadly, with continued contributions by Kemp. Significantly, Kemp and his colleagues early-on developed commercially-available equipment. Used in the default settings, the equipment was good for hearing screening. But those who wanted to dig deeper could set up solid clinical-research protocols. This allowed translational and clinical research to progress alongside basic research. While OAEs are most often used clinically as pass/fail hearing-screening or site-of-lesion tests, Kemp at the outset inspired other uses, including detection of cochlear changes. He laid out critically important principles that have served as cornerstones for applied OAE research in the area of noise exposure.

2:25

2pPPa5. David Kemp's impact on cochlear modeling and otoacoustic emission measurement and modeling. Carrick L. Talmadge (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38655, clt@olemiss.edu) and Glenis R. Long (Speech-Language-Hearing Sci., Graduate Ctr. CUNY, New York, NY)

Between 1990 and 1998, Drs. Carrick Talmadge, Glenis Long and Arnold Tubis worked together in the field of Hearing Science at Purdue University on a diverse number of topics, including the measurement and modeling of spontaneous otoacoustic emissions, measurement and modeling of stimulus frequency and distortion product otoacoustic emissions (SFOAEs and DPOAEs respectively), and on cochlear modeling. We will explore here linkages between Dr. Kemp's work and our research. This includes his sweep method for measurement of SFOAEs, which lead to the least-squares-fit based method for recording and analyzing SFOAEs and DPOAEs developed by our group. His emphasis on the clinical relevance of otoacoustic emissions implied a linkage between otoacoustic emission data and the underlying function of the cochlea, and this became a main focus of our cochlear modeling efforts. His evidence for two-sources for DPOAEs and his study of the so-called DPOAE "filter shape" were highly influential in the direction of our research at Purdue University.

2:45–3:00 Break

3:00

2pPPa6. Interrelationships among microstructures of otoacoustic emissions and hearing thresholds. James Dewey (Dept. of Otolaryngol. - Head & Neck Surgery, Univ. of Southern California, 1501 San Pablo St., ZNI Rm. 407, Los Angeles, CA 90033, james.dewey@med.usc.edu) and Sumitrajit Dhar (Roxelyn & Richard Pepper Dept. of Commun. Sci. & Disord., Northwestern Univ., Evanston, IL)

Otoacoustic emission (OAE) amplitudes, phases, and delays often exhibit ripples with small changes in stimulus frequency. Similar microstructure patterns are observed in behavioral hearing thresholds and loudness judgements for pure tones. Here, we summarize work demonstrating that there is a common microstructure in hearing thresholds and OAEs elicited by single tones—referred to as stimulus-frequency OAEs (SFOAEs)—when presented at near-threshold levels. The periodicity and strength of this microstructure are related to SFOAE delay and magnitude, respectively. Such relationships are consistent with this microstructure arising from multiple intracochlear reflections of SFOAE energy between its region of generation and the middle ear boundary—an idea proposed by David Kemp in his first reports describing the discovery of emissions from the ear. The details of OAE generation and propagation explain why similar but not identical microstructures are also observed in the ear canal pressure in response to tones (which is the vector sum of the stimulus and SFOAE pressures) and in two-tone evoked distortion-product OAE spectra, as well as why microstructures shift in frequency as the result of cochlear or middle ear manipulations.

3:20

2pPPa7. Temporal dynamics of the generator of stimulated otoacoustic emissions. Sarah Verhulst (Dept. Information Technol., Ghent Univ., Technologiepark 15, Zwijnaarde 9052, Belgium, s.verhulst@ugent.be), Alessandro Altoè (Dept. of Signal Processing and Acoust., Aalto Univ., Aalto, Finland), Stefan Raufer (Speech and Hearing BioSci. and Technol. Program, Harvard Univ., Boston, MA), Karolina Charaziak, and Christopher Shera (Caruso Dept. of Otolaryngol., Univ. of Southern California, Los Angeles, CA)

The early emission work of David Kemp investigated temporal properties of click-evoked otoacoustic emissions (CEOAEs). Suppressor clicks were positioned up to 10 ms before or after the evoking (test) click and reduced the emission amplitude. Intriguingly, suppressors that preceded the test click by 1-2 ms were more effective than simultaneously presented suppressors. This observation seems not to support the hypothesis that emission generation operates on the basis of an instantaneous gain/suppression mechanism, for which simultaneously presented suppressors should be most effective. Kemps' observations thus left the field with an important OAE generation question that inspired several OAE studies. Kemps' results can be explained on the basis of (i) a non-instantaneous gain mechanism at the OAE generation site, or (ii), complex temporal interactions of basilar-membrane impulse responses (BM IRs) that operate with an instantaneous nonlinearity, but yield the observed non-instantaneous suppression properties. We investigated these dynamics using a nonlinear transmission-line model of the human cochlea and found that maximal CEOAE suppression can occur for preceding clicks even when the cochlear nonlinearity is kept time-invariant, supporting hypothesis (ii). Additionally, we used the frequency-dependence of CEOAE suppression to quantify human BM IR duration and cochlear filter tuning, yielding Q_{ERB} estimates of 13.8 F[in kHz]$^{0.22}$.

3:40

2pPPa8. Intracochlear evidence on generation and propagation of distortion product otoacoustic emissions in gerbil cochlea. Wei Dong (ENT, Loma Linda Univ. Health, 11201 Benton St., Loma Linda, CA 92374, wdong@llu.edu)

Otoacoustic emissions (OAEs, Kemp, 1978) are complex signals appearing as a sum of multiple components relating to different cochlear source regions and different active mechanisms thus making it difficult to ascribe them to specific hearing frequencies. Improving the utility of OAEs requires understanding the origin of these multiple components and extracting the information they carry to the ear canal. The current presentation aims to provide direct intracochlear observations on OAE generation and their transmission. Two- or three-tone evoked pressure responses were simultaneously measured in the gerbil ear canal and scala tympani. Distortion product otoacoustic emissions (DPOAEs) and their intracochlear sources as DPs were analyzed. Results demonstrated: 1) DPOAEs come mainly from the f_2 peak region, and extend further basal with increases in primary-tone intensities. These findings are consistent with several notions including: the theoretical prediction of "wave-fixed" or nonlinear generation, and the location of cochlear amplifier; and 2) DPs post-generation may travel both forward to their own best frequency places and in reverse to the stapes. The reverse propagation of DPs was mainly through reverse traveling waves, even though compression waves may also contribute to DPOAE generation, especially, to those produced close to the stapes in the basal cochlear region.

4:00

2pPPa9. Comparative otoacoustic insights. Christopher Bergevin (Phys. & Astronomy, York Univ., 4700 Keele St., Petrie 240, Toronto, ON M3J 1P3, Canada, cberge@yorku.ca)

"Cochlea wave propagation…" These words started off David Kemp's seminal 1978 contribution to auditory biophysics, where he first described "a new auditory phenomenon" [JASA 64(5):1386-1391]. The impact of this report has transcended numerous disciplines, finding an impressive range of scientific and clinical applications (e.g., monitoring cochlear status to ototoxicity, biophysical models of the cochlea, pediatric hearing screening). One area where this is most readily apparent is the "comparative" one: It appears that animals all across the phylogenetic tree exhibit robust OAE activity. Given broad morphological differences (e.g., cochlear length, presence/absence of a flexible basilar membrane and/or stereovillar hair cells), a comparative otoacoustic approach offers remarkable potential for identifying universal biophysical principles at play. This talk will compare/contrast OAE properties (and the associated implications) across a broad range of groups such as: humans, birds, frogs, lizards, ferrets, non-human primates, tigers, and even insects. Furthermore, by examining OAEs in species where the traditional basilar membrane traveling wave is absent (e.g., reptiles), we stand to gain biophysical insight into the precise role of "wave propagation." Taken together, the enormous impact the otoacoustic approach has had towards understanding the biomechanics of the inner ear speaks to the truly profound nature of the "Kempian" discovery.

Contributed Paper

4:20

2pPPa10. The role of the tectorial membrane in cochlear mechanics. Karl Grosh, Amir Nankali, and Aritra Sasmal (Univ. of Michigan, 2350 Hayward St., Dept. of Mech. Eng., Ann Arbor, MI 48109-2125, grosh@umich.edu)

The tectorial membrane (TM) is strategically located in the mammalian cochlea, anchoring the apical pole of the outer hair cell stereocilia and hovering just above the sensory inner hair cell stereocilia. Genetic mutations of TM-specific proteins cause nonsyndromic deafness in humans while experiments in mutant mice clearly demonstrate the impact that manipulations of TM structural proteins have on hearing thresholds and cochlear sensitivity. This direct evidence shows the importance of this cochlear structure. Mathematical models (with Zwsilocki as an early proponent) have long incorporated the TM but have not included a complete electrical and fluidic coupling. We show that an electromechanical model of the active processes in the cochlea combined with a fluid-structure model that strongly couples the basilar membrane and the TM to the cochlear fluids is crucial in understanding the low and high frequency response in a base-to-apex model of the cochlea. Specifically, we show that the onset of nonlinearity at high frequencies is due to a radial resonance of the uncoupled TM and investigate the influence of fluid-loading on the TM and TM viscoelasticity on the low-pass behavior seen in the apex. A connection of this new wave bearing mechanism to otoacoustic emissions will be discussed.

4:35–4:55 Panel Discussion

Session 2pPPb

Psychological and Physiological Acoustics: Binaural Hearing and Psychoacoustic Methodology (Poster Session)

Yonghee Oh, Chair
Otolaryngology-Head and Neck Surgery, Oregon Health & Science University, Mailcode NRC04, 3181 SW Sam Jackson Park Road, Portland, OR 97239

All posters will be on display from 1:00 p.m. to 5:00 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 1:00 p.m. to 3:00 p.m. and authors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

2pPPb1. The acoustic images in the fields of knowledge. Peter Francis C. Gossweiler (Instituto de Artes, UFRGS, Trav. Nsa. Sra. de Lourdes, 230 apt. 503 D, Porto Alegre, Rio Grande do Sul 91920-040, Brazil, petergossweiler@gmail.com)

Can we provide different acoustic images, ontologically speaking, from the fields of knowledge? As an engineer that experiences acoustic by processing mathematical calculations (with an evaluative approach), an architect by projecting physical spaces (focusing on audible comfort), an artist by proposing sound enigmas and mind gaps (perhaps proposing an audible discomfort), a musician by dealing with the reflections of his sound making (awaking the pleasure of listening), a religious man by preaching spiritual rituals (addressing a meditative state for his assembled group of people), just to name a few. Thus, our research evokes philosophers such as Martin Heidegger (by the concept of dasein), Michel Certeau (which suggests exercising rebellion in everyday life) and Marie-José Mondzain (with the birth of the homo-spectator, by his desire for images) to point out, with those key operative concepts, outstanding approaches to how we can investigate the internal site specifics of the human being, in his dynamic and unfinished possibilities, the variables of the listening regime on acoustic images in the fields of knowledge.

2pPPb2. Observations on the shape of hearing sensations in pure-tone lateralization. Florian Völk (WindAcoust., Muehlbachstrasse 1, WinDC 86949, Germany, voelk@windacoustics.com), Jörg Encke, Diana Reimann, and Werner Hemmert (Bio-Inspired Information Processing, Tech. Univ. of Munich, Garching, Germany)

In certain conditions, especially with diotic headphone presentation, hearing sensations are located inside the head. Dichotic headphone presentation can result in lateralization: delaying one of the headphone input signals or reducing its amplitude typically pushes the hearing sensation inside the head towards the contralateral ear. Systematic connections between physical stimulus parameters and hearing-sensation properties provide insight into auditory-localization mechanisms and are helpful in designing and evaluating models thereof. While various studies on the lateral displacement of hearing sensations exist, fewer data is available on the respective shape or spatial extent. Based on observations from a pure-tone lateralization experiment, this study aims at taking a first step towards quantifying the typical shape of the corresponding hearing sensations: in two separate experiments, twelve normal-hearing subjects reported a) the overall spatial extent of the hearing sensation or, if applicable, of all simultaneously occurring sensations and b) the number of simultaneously occurring, spatially separated hearing sensations. The data suggests, in line with earlier studies, considerable differences between the results for pure interaural phase and amplitude differences, respectively. Based on the results, an initial empirical description of the stimulus-dependent hearing sensations in pure-tone lateralization summarizes the observations, especially regarding location and spatial shape.

2pPPb3. A real-time method for computing confidence intervals for trial-based psychoacoustic tests. Jakob Hartl (NASA Interns, Fellows & Scholars, Hampton, VA) and Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, andrew.christian@nasa.gov)

The vast majority of psychoacoustic testing involves presenting a subject with a series of short, simple questions (trials). A common desire is to track the level of statistical confidence in the results after each trial so that one can stop the test once a threshold of certainty has been surpassed. Typically, this certainty statistic will take the form of a confidence interval (CI). Past methods to compute CIs involve assumptions that may impact the performance of the resultant test-stopping criterion. This talk proposes a method to compute CIs that is free of assumptions regarding: the testing procedure used (e.g., up-down staircase, PEST, non-adaptive), the testing modality used (e.g., n-alternative forced-choice, yes/no), and the model psychometric function used (e.g., logit, Gompertz). The method involves a fixed-quadrature integration of the likelihood function of the model parameters given the data retrieved after each trial. The execution time is demonstrated to be low enough so that the method may run during the inter-trial pause. The method is demonstrated for a 3-alternative forced-choice detection task involving two interleaved transformed up-down staircases (between which the data is pooled) using a scaled normal CDF as the model for the psychometric function.

2pPPb4. Localization of adventitious respiratory sounds. Brian Henry and Thomas J. Royston (BioEng., Univ. of Illinois at Chicago, 851 South Morgan St., MC 063, Chicago, IL 60607, troyston@uic.edu)

In a recent publication [Henry and Royston, J. Acoust. Soc. Am., 142, 1774–1783 (2017)], an algorithm was introduced to calculate the acoustic response to externally introduced and endogenous respiratory sounds within a realistic, patient-specific subglottal airway tree. This work is extended using an efficient numerical boundary element (BE) approach to calculate the resulting radiated sound field from the airway tree into the lung parenchyma taking into account the surrounding chest wall. Within the BE model of the left lung parenchyma, comprised of more than 6,000 triangular surface elements, more than 30,000 monopoles are used to approximate complex airway-originated acoustic sources. The chest wall is modeled as a boundary condition on the parenchymal surface. Several cases were simulated, including a bronchoconstricted lung that had an internal acoustic source introduced in a bronchiole, approximating a wheeze. An acoustic source localization algorithm coupled to the BE model estimated the wheeze source location to within a few millimeters based solely on the acoustic field at the surface. Improved noninvasive means of locating adventitious respiratory sounds may enhance our understanding of acoustic changes correlated to pathology, and potentially provide improved noninvasive tools for the diagnosis of pulmonary diseases that uniquely alter acoustics.

2pPPb5. Binaural perception of stereo noise bursts as a function of burst duration and degree of interchannel coherence. Steven E. Crawford and Mark F. Bocko (Elec. Eng., Univ. of Rochester, 500 Joseph C. Wilson Blvd., Rochester, NY 14627, steven.crawford@rochester.edu)

Amplitude panning for virtual sound source rendering in stereo and multi-channel audio reproduction is well established. Binaural fusion can be represented by vector addition of the acoustic wave propagation vectors from individual speakers; however, this applies only to stereo signals with perfect interchannel coherence and does not provide a description of the perceived sound image when the interchannel coherence is less than complete. Listening tests show that the stereo image formed by continuous incoherent noise fills the space between the loudspeakers; however, for a single short noise burst (a few milliseconds long) the virtual source collapses to a single point in space with its location determined by the position of the maximum of the interchannel correlation function. Concatenation of an ensemble of short noise bursts with a distribution of interchannel correlation function maxima creates a long noise burst displaying a broad peak in its interchannel correlation function; the location and width of which correspond to the apparent location and spread of the virtual sound image. We present a simple model that combines the effective averaging time of the human central binaural fusion mechanism and the interchannel coherence of the stereo signal to predict the spatial properties of virtual sound sources.

2pPPb6. Vision-induced reweighting of binaural localization cues. Maike Ferber, Bernhard Laback (Acoust. Res. Inst., Austrian Acad. of Sci., Wohllebengasse 12-14, Vienna 1040, Austria, mferber@kfs.oeaw.ac.at), and Norbert Kopco (Pavol Jozef Safarik Univ., Kosice, Slovakia)

Normal-hearing listeners apply frequency-dependent weights when combining the two binaural cues (interaural time and level difference, ITD and ILD) to determine the perceived sound source azimuth. Cochlear-implant (CI) listeners, however, rely almost entirely on ILDs. Since current CI systems do not reliably convey ITD information, CI listeners might learn to ignore ITDs and focus on ILDs instead. We investigated whether this reweighting of binaural cues is generally possible. 20 normal-hearing participants, assigned to two groups, completed an experiment in a virtual audio-visual environment. The experiment consisted of a pre-test to establish the initial ITD/ILD weights, a seven-day training, in which visual feedback reinforced one of the cues, and a post-test to measure the effect of training on the weights. Participants' task was to lateralize octave-wide bands of noise (centered at 2.8 kHz) containing various combinations of spatially inconsistent ITD and ILD. In both groups, the lateralization bias related to the reinforced cue declined significantly from pre- to post-test, suggesting that participants reweighted the binaural cues in accordance with the visual feedback. These results are promising in terms of making ITD information usable by CI listeners once it is conveyed by the implants. [Support: SpaCI Danube Partnership project, H2020-MSCA-RISE-2015 #691229.]

2pPPb7. The effect of visual representation of a room on perceived reverberation. Michael Schutte (Div. of Neurobiology, Dept. Biology II, Ludwig-Maximilians-Universität München, Großhaderner Straße 2, Planegg-Martinsried 82152, Germany, michael.schutte@lmu.de), Stephan D. Ewert (Medizinische Physik und Exzellenzcluster Hearing4all, Universität Oldenburg, Oldenburg, Germany), and Lutz Wiegrebe (Div. of Neurobiology, Dept. Biology II, Ludwig-Maximilians-Universität München, Munich, Germany)

In everyday environments, sound reflections from walls and other objects arrive at the listener's ear in addition to the direct sound radiated from a source. Humans have evolved mechanisms to perceptually suppress distracting early reflections, summarized as the precedence effect. Visual information of the surroundings influences the effectiveness of this mechanism. It has also been shown that similar compensation effects can occur for later reflections and reverberation, and that humans can estimate acoustic properties of a room from the visual impression. Taking these findings together, we hypothesize that the visual impression of a room affects the perception of its reverberation. The hypothesis was tested in a highly immersive audio-visual virtual reality environment consisting of a horizontal loudspeaker array in an acoustically treated chamber and a head-mounted display. In a magnitude estimation paradigm, subjects judged the perceived degree of reverberation in conditions where the simultaneously presented acoustic and visual stimuli were either matched regarding the room, sound source azimuth, and sound source distance, or diverged in one of those aspects. Audio-only control conditions served as a baseline. Preliminary results from six subjects suggest a predominant reliance on auditory input when quantifying perceived reverberation even in audio-visually conflicting conditions.

2pPPb8. Toward a visual assistive listening device: Real-time synthesis of a virtual talking face from acoustic speech using deep neural networks. Lele Chen (Dept. of Comput. Sci., Univ. of Rochester, Rochester, NY), Emre Eskimez (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY), Zhiheng Li (Dept. of Comput. Sci., Univ. of Rochester, Rochester, NY), Zhiyao Duan (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY), Chenliang Xu (Dept. of Comput. Sci., Univ. of Rochester, Rochester, NY), and Ross K. Maddox (Departments of Biomedical Eng. and Neurosci., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rm. 5.7425, Rochester, NY 14642, ross.maddox@rochester.edu)

Speech perception is a crucial function of the human auditory system, but speech is not only an acoustic signal—visual cues from a talker's face and articulators (lips, teeth, and tongue) carry considerable linguistic information. These cues offer substantial and important improvements to speech comprehension when the acoustic signal suffers degradations like background noise or impaired hearing. However, useful visual cues are not always available, such as when talking on the phone or listening to a podcast. We are developing a system for generating a realistic speaking face from speech audio input. The system uses novel deep neural networks trained on a large audio-visual speech corpus. It is designed to run in real time so that it can be used as an assistive listening device. Previous systems have shown improvements in speech perception only for the most degraded speech. Our design differs notably from earlier ones in that it does not use a language model—instead, it makes a direct transformation from speech audio to face video. This allows the temporal coherence between the acoustic and visual modalities to be preserved, which has been shown to be crucial to cross-modal perceptual binding.

2pPPb9. Effects of vision, listener head movement, and target location on spatial selective auditory attention. Ewan A. Macpherson 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. We have reported [Macpherson & Ellis, ASA Salt Lake, 2016] that listeners' ability to attend to a frontal target in the presence of spatially separated distractors is decreased in a head-movement condition, suggesting that listeners cannot rapidly update the focus of their SSAA to compensate fully for head motion. Participants reported anecdotally that being able to see the target loudspeaker seemed beneficial in directing SSAA. Here, we assessed the benefit of access to a visual reference frame in a similar task. Under lighted and dark conditions, on each trial, listeners either fixated toward 0 azimuth or oscillated their heads at ~0.5 Hz with an amplitude of ~$\pm$40 degrees while five different simultaneous equal-intensity sequences of four spoken digits were presented as target (at azimuths from 0 to 90 degrees) and four distractors (at azimuths $\pm$22.5 and $\pm$45 degrees relative to the central target). Listeners verbally reported the target sequence heard. Performance declined under head motion and with increasing target eccentricity, but not additionally in dark conditions, suggesting that a visual reference frame was not beneficial in directing SSAA in our task.

2pPPb10. Audiovisual speech detection benefits: A comparison across psychophysical approaches. Kaylah Lalonde (Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68131, kaylah.lalonde@boystown.org) and Lynne A. Werner (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

Visual speech reduces uncertainty as to when auditory speech will occur, thereby enhancing detection of speech in noise. These detection benefits reflect a low-level perceptual mechanism of AV speech enhancement that is independent of the linguistic nature of speech. Audiovisual speech detection benefits have primarily been assessed using two-interval forced-choice paradigms. Results from AV speech detection experiments using three different psychophysical approaches will be presented. These include a single-interval yes/no task, a go/no-go task, and a two-interval forced choice task. In all experiments, adults with normal hearing complete an auditory-only condition with a still image of the talker and an audiovisual condition with a synchronous and congruent visual speech signal. The audiovisual condition includes foil trials/intervals that force participants to use the auditory information to make the decision. Adults demonstrate significant audiovisual benefit to speech detection across all three task, but benefit on the yes/no task depended on the analysis method. Differences in benefit across tasks and signal-to-noise ratios will be discussed. Results of ongoing and planned experiments that aim to isolate the contribution of particular visual temporal cues (onsets, amplitude envelope) will also be presented. [Work supported by NIDCD R01 DC000396, F32 DC015387, and T32 DC005361.]

2pPPb11. Role of expectancy in front-back confusions during sound source localization. William Yost (ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Front-back confusions (FBCs) often occur in laboratory experiments. Stimulus complexity, especially for high-frequency sounds, and head motion significantly reduce FBCs, so that in the everyday world FBCs probably do not often influence sound source localization. There is a small literature suggesting that listeners "expectancies" (based on experience or other sensory inputs) for where sound sources might be also affect FBCs. A low-pass noise that produces a large number of FBCs and a high-pass noise that does not were used. Listeners indicated the location of sound sources on the azimuth plane. The study aimed to investigate FBCs and not sound source localization accuracy. For one condition, listeners were told that the noises would all be coming from in front, that in another condition from in back, and in a third condition from the full 360 azimuth circle. In the first (front) and second (back) conditions, most of the sounds came from the front or back respectfully, but 20% did not. The expectancy that sounds would be in front or in back had a large influence on FBCs as compared to when sounds were presented from the entire azimuth circle. (Work supported by grants from NIDCD and Oculus VR, LLC.)

2pPPb12. Modeling binaural detection of a Gaussian noise target in the presence of a lead/lag masker. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu), M. Torben Pastore (Speech and Hearing Sci., Arizona State Univ., Troy, New York), and Yi Zhou (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Recently, we presented the results of a binaural masked detection experiment in which a noise target was temporally embedded within a lead/lag noise masker pair [J. Acoust. Soc. Am. 141, 3639]. The results show that the inter-stimulus interval (ISI) between the masker and its reflection changed the detection threshold significantly. For low ISIs of 2-ms, the average masked detection was -11 dB, but for a greater ISI of 20 ms, the threshold was much higher (-2 dB). In the experiment, masked detection thresholds did not depend on whether the masker lead was on the same side as the target (with the lag on the contralateral side) or the other way round. Different model approaches are presented to simulate the psychophysical results, including a precedence effect (PE) model that was extended using a cepstrum-based method to determine the localization cues for the direct sound of the masker. The PE model is used in conjunction with an EC model and a binaural/monaural detector using optimal frequency-weighting functions that were calculated during the above-threshold conditions during the adaptive experiment. The modeling results show that the PE model is not needed to explain the psychoacoustical results. [Work supported by NSF BCS-1539276 and NSF BCS-1539376.]

2pPPb13. A correlation-based binaural localization model with reflection identification using neural networks. Nikhil Deshpande and Jonas Braasch (Architecture, Rensselaer Polytechnic Inst., 220 3rd St., Troy, NY 12180, deshpn@rpi.edu)

This model identifies a lateralized direct sound and single reflection. Anechoic speech is lateralized using a head-related transfer function. The speech is then time-delayed and lateralized to a different angle. The model then utilizes a binaurally-integrated cross-correlation/auto-correlation mechanism (BICAM) to analyze the lead/lag stimulus and generate a band-limited binaural activity pattern. This output is analyzed to calculate the time delay of the reflection, and the model then uses a neural network to estimate the lateralization of both the direct and reflected sound. From here, the model can remove the reflected copy of the sound, and extract the original raw, anechoic speech. [This work was supported by the HASS Fellowship at Rensselaer Polytechnic Institute, and the National Science Foundation Grant No. NSF BCS-1539276.]

2pPPb14. A comparison of free-field and headphone based sound localization tasks. Devon Brichetto, Brock Carlson, Mary Landis Gaston, Thomas Olson (Neurosci., St. Olaf College, 1520 St. Olaf Ave., Northfield, MN 55057, briche1@stolaf.edu), and Jeremy Loebach (Psych., St. Olaf College, Northfield, MN)

Accurate sound localization is critical for identifying the origin of a sound source and for segregating sources in complex auditory scenes. Sound localization tasks often use free-field presentation of stimuli from a speaker array, or synthetic signals presented over headphones. This study compares localization of ITD and ILD cues presented in the free-field using the SoLo-Arc (Westerberg *et al.*, 2015), a custom designed speaker array that can systematically present auditory and visual stimuli in a 180-degree free-field environment in both the horizontal and vertical planes, with synthetic ITD and ILD cues presented via headphones. Further, we compared two ways for participants to indicate their response: a manual system where participants pointed a laser at the source, which was identified by a spotter, and an automated potentiometer coupled servo system where the voltage derived from the potentiometer is converted into angular location, allowing the participant to indicate source localization without movement of the head or body. Results indicated that localization in the free-field tasks was significantly more accurate than the headphone tasks for both ITD and ILD cues, and that the automated pointing system was superior to the manual system. Pilot data from the elevation task will also be discussed.

2pPPb15. Portable Automated Rapid Testing (PART) measures for auditory research. Frederick J. Gallun (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov), Aaron Seitz (Psych., Univ. of California, Riverside, Riverside, CA), David A. Eddins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), Michelle R. Molis (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR), Trevor Stavropoulos (Psych., Univ. of California, Riverside, Riverside, CA), Kasey Jakien, Sean Kampel (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR), Anna C. Diedesch (Commun. Sci. and Disord., Western Washington Univ., Bellingham, WA), Eric Hoover, Karen Bell (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), Pamela Souza, Melissa Sherman (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL), Lauren Calandruccio, Gretchen Xue (Psychol. Sci., Case Western Univ., Cleveland, OH), Rene Sebena (Psych., Pavol Jozef Safarik Univ., Kosice, Slovakia), and Nirmal Kumar Srinivasan (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., Towson, MD)

This presentation will focus on data collected across the globe on tablet computers using applications available for free download, built-in sound hardware, and calibrated consumer-grade headphones. Tests involve spatial

release from speech-on-speech masking, binaural sensitivity, gap discrimination, temporal modulation, spectral modulation, and spectrotemporal modulation. The data will be compared across sites and with data from the published literature. The similarity of the obtained data to the expected values and the consistency across sites will help evaluate the potential of this relatively inexpensive and easily disseminated approach to psychoacoustical data collection. The extent to which valid performance can be obtained is a metric of the possibility that in the future such test methods could be used by researchers without access to a full laboratory, clinicians interested in evaluating auditory function beyond the audiogram, and students as part of their training, as well as many other uses not yet imagined.

2pPPb16. A hearing test simulator GUI for clinical testing. Serkan Tokgoz, Yiya Hao, and Issa M. Panahi (Dept. of Elec. Eng., The Univ. of Texas at Dallas, EC33 University of Texas at Dallas 800 West Campbell Rd., Richardson, TX 75080, sxt167830@utdallas.edu)

This paper presents an overview of a useful MATLAB based GUI for hearing testing to evaluate the subjects' hearing ability in noisy environments with different SNR values. With this software package, the examiner will be able to identify which words are correctly perceived and then collect test data in various conditions to measure the performance of the subjects in hearing tests. From the subjects' responses, word recognition rates are saved by the examiner with different noise types such as babble, traffic, machinery, and white noise. Additionally, the subjective tests are completed through repeated testing circles. All hearing testing data and word recognition rate are saved into the database for later use purpose and analysis. This computer aided simulation makes a reliable and cost effective to create real environmental conditions for clinical testing. Our MATLAB based GUI addresses the needs of both clinical evaluation and engineering.

2pPPb17. The effects of interaural level differences on binaural fusion in normal-hearing listeners. Bess Glickman, Yonghee Oh, and Lina Reiss (Otolaryngology-Head and Neck Surgery, Oregon Health and Sci. Univ., 3181 SW Sam Jackson Park Rd., Mail Code: NRC04, Portland, OR 97239, glickman@ohsu.edu)

Sound localization is an important aspect of auditory scene analysis, allowing listeners to group acoustic components from the same location into a single stream (Bregman 1990). Binaural pitch fusion, the fusion of different frequency tones across ears, can be thought of as one type of auditory streaming. Little is known about how sound localization cues affect binaural fusion. The goal of this study was to investigate the effects of interaural level differences (ILDs), one cue for sound localization, on binaural fusion of dichotic tones. Binaural pitch fusion was measured in adult normal-hearing (NH) listeners, using five ILD conditions: ILD = 0, 1.25, 2.5, 5, and 10 dB. Fusion ranges were measured by simultaneous presentation of reference and comparison stimuli in opposite ears, and varying the comparison stimulus to find the frequency range that fused with the reference stimulus. Preliminary results (5 NH; data collection is ongoing) show that small ILDs increase fusion ranges, while larger ILDs decrease fusion ranges. These findings suggest that ILDs can affect fusion range in NH listeners, and imply that simulated sound source location may provide a top-down grouping cue for binaural fusion. [This research was funded by a NIH-NIDCD grant R01 DC013307.]

2pPPb18. Is there an auditory identification "pop-out" for moving speech targets? M. Torben Pastore (Speech and Hearing Sci., Arizona State Univ., 4 Irving Pl., Troy, New York 12180, m.torben.pastore@gmail.com) and William Yost (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Stationary visual targets are often far more salient when the target moves against an otherwise static background—the so-called "pop out" effect. In the first of two experiments, we tested for a similar effect in the auditory domain. Listeners were asked to identify a single word, spoken by a female, in the midst of two or four masking words spoken by males. Listener performance was estimated in terms of percentage of correct responses and compared between conditions where target and maskers were co-located or located at different locations. For some conditions, the target word was amplitude-panned across the loudspeaker array and in others the target remained stationary. Results showed a spatial release from masking for all conditions where the target and maskers were at different locations. The presentation of a stationary versus moving target stimulus yielded no statistically significant difference in identification performance, suggesting that the visual "pop-out" effect may not have a direct corollary in the auditory domain.

2pPPb19. Is there an auditory detection "pop-out" for moving tones in the presence of static noise maskers? M. Torben Pastore (Speech and Hearing Sci., Arizona State Univ., 4 Irving Pl., Troy, New York 12180, m.torben.pastore@gmail.com) and William Yost (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Stationary visual targets often become far more salient when they move against an otherwise static background—the so-called "pop out" effect. In the second of two experiments, we tested for a similar pop-out effect in the auditory domain. Tone-in-noise detection thresholds were measured using a 2-down 1-up adaptive procedure under conditions where the target was stationary or moved via amplitude-panning. Target frequencies of 500 Hz and 4 kHz were tested. Maskers (2–4, depending on the condition) were independent Gaussian noises filtered to have equal energy per octave. All target and masker stimuli were 500-ms duration. Listener performance was compared between conditions where target and maskers were co-located or positioned separately for conditions when the target word was amplitude-panned across the loudspeaker array and in others when the target remained stationary. Results will be presented and discussed.

2pPPb20. Effect of auditory stream segregation cues on binaural pitch fusion. Yonghee Oh, Frederick J. Gallun, and Lina Reiss (Otolaryngology-Head and Neck Surgery, Oregon Health & Sci. Univ., Mailcode NRC04, 3181 SW Sam Jackson Park Rd., Portland, OR 97239, oyo@ohsu.edu)

Binaural pitch fusion, the fusion of dichotically presented tones that evoke different pitches across ears, can be influenced by temporal envelope cues such as coherent amplitude modulation (Oh and Reiss, ASA 2017). This suggests that binaural pitch fusion may be governed at least in part by top-down processes involved in auditory grouping. The current study was designed to investigate how temporal streaming cues either prior or posterior to a fused dichotic tone pair can influence binaural pitch fusion. Six normal-hearing (NH) listeners and six bilateral hearing-aid (HA) users were tested in a modified auditory streaming paradigm (Steiger and Bregman, 1982). A pair of simultaneous, dichotic reference and comparison stimuli was placed in rapid alternation with a third binaural (diotic) capture stimulus, for a total of 10 alternations. The binaural fusion ranges, the frequency ranges over which binaural pitch fusion occurred for dichotic stimuli, were smaller when measured with the streaming paradigm than with static, spectral-only stimuli of equivalent durations. HA users showed greater reductions in fusion ranges with streaming than NH listeners. The findings suggest that temporally flanking stimuli suppress fusion between dichotic stimuli. [Work supported by NIH-NIDCD grant R01 DC013307 and F32 DC016193.]

2pPPb21. Assessing cognitive and perceptual abilities by manipulating task requirements when listening to speech in noise. Rachel E. Miller and Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, 1229 Marion St, Columbia, SC 29201, remiller@email.sc.edu)

Speech recognition involves both cognitive and perceptual processing that can be differentially affected by the task requirements. This study assessed the degree to which different cognitive and perceptual abilities explained speech recognition using the Coordinate Response Measure (CRM). Speech recognition was assessed by varying the requirements of attention (i.e., selective or divided attention), and by assessing perceptual processes related to speech glimpsing (i.e., steady-state noise versus speech-modulated noise). Additional cognitive and perceptual tasks assessed cognitive interference, working memory, and amplitude modulation detection at 4 and 10 Hz. Results demonstrated that masking release (was most associated with modulation detection, indicating that the ability to take advantage of momentary dips in the masker (i.e., glimpsing) requires better amplitude

modulation detection abilities. In contrast, when attentional constraints were imposed, a measure of cognitive interference was most predictive of performance for reporting keywords spoken by both the target talker (i.e., accurate responses) and competitor (i.e., intrusion responses). Interestingly, our measure of working memory was not correlated with any of the CRM tasks. These results suggest that conditions in which the CRM task is administered (i.e. manipulating noise or attention) can differentially weight separate perceptual and cognitive skills. [Work supported, in part, by NIH/NIDCD.]

2pPPb22. Listening to degraded speech can cause listeners to "wait and see". Steven P. Gianakas and Matthew Winn (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St, Seattle, WA 98105, spgia5@uw.edu)

The "wait and see" approach is described as waiting until hearing the end of a sentence or word before committing to a response. Although people with normal hearing (NH) perceive speech in a very rapid incremental fashion, listeners with cochlear implants (CIs) appear to utilize the wait and see approach in tests of word recognition. Specifically, they not only present with lower accuracy, but do not respond (even with eye gaze) until the utterance is fully or nearly complete. This implies that in addition to poorer accuracy scores, CI listeners might be adopting a different, more reluctant approach to word recognition. To better understand why listeners may adopt a reluctant listening strategy, we examined word recognition with a spectrally degraded speech signal in NH listeners, where eye gaze was tracked as a real-time indicator of commitment to a response. Preliminary results show a decrease in accuracy and a delay in response time for NH listeners when the signal is vocoded. However, the delays were not as extreme as those observed in listeners with CIs. These findings would suggest listeners hearing in a severely degraded condition may change their approach to listening, but only after extensive experience adjusting to the signal.

2pPPb23. Differential effects of hearing loss on neural and behavioral measures of speech perception in noise. Tess K. Koerner (VA RR&D National Ctr. for Rehabilitative Auditory Res., 3125 Holmes Ave. South #204, Minneapolis, WI 55408, koern030@umn.edu) and Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Understanding speech in background noise is difficult for many listeners, and those with hearing impairment tend to show considerable variability in performance. Event-related potentials (ERPs) and associated cortical oscillations are useful for examining neural responses to speech in various listening conditions and may represent neurophysiological markers of auditory perception. The present study examined the effects of hearing impairment on the neural coding and perception of speech in noise in a group of adult listeners with varying degrees of sensorineural hearing loss. This work was also designed to determine whether cortical ERPs and oscillatory rhythms in various frequency bands can predict the effects of hearing impairment on sentence recognition in noise. Passive N1-P2 and MMN responses were elicited with a double-oddball paradigm containing a consonant and vowel change in background noise. Speech perception was evaluated using phoneme discrimination and sentence recognition in noise tasks. Analysis showed that hearing impairment significantly impacted the MMN response but not the obligatory, sensory N1-P2 complex. Results showed that these objective neural responses may be promising neurophysiological markers of perception in noise across listeners. This work has important implications regarding the use of electrophysiological measures for assessing speech perception in complex auditory environments in clinical populations.

2pPPb24. Human recognition of environmental sounds is not always robust to reverberation. James Traer and Josh McDermott (Brain and Cognit. Sci., MIT, 77 Massachusetts Ave., Cambridge, MA 02139, jtraer@mit.edu)

Reverberation is ubiquitous in natural environments, but its effect on the recognition of non-speech sounds is poorly documented. To evaluate human robustness to reverberation, we measured its effect on the recognizability of everyday sounds. Listeners identified a diverse set of recorded environmental sounds (footsteps, animal vocalizations, vehicles moving, hammering, etc.) in an open set recognition task. For each participant, half of the sounds (randomly assigned) were presented in reverberation. We found the effect of reverberation to depend on the typical listening conditions for a sound. Sounds that are typically loud and heard in indoor environments, and which thus should often be accompanied by reverberation, were recognized robustly, with only a small impairment for reverberant conditions. In contrast, sounds that are either typically quiet or typically heard outdoors, for which reverberation should be less pronounced, produced a large recognition decrement in reverberation. These results demonstrate that humans can be remarkably robust to the distortion induced by reverberation, but that this robustness disappears when the reverberation is not consistent with the expected source properties. The results are consistent with the idea that listeners perceptually separate sound sources from reverberation, constrained by the likelihood of source-environment pairings.

Session 2pSA

Structural Acoustics and Vibration and Education in Acoustics: Improving Education in Structural Acoustics and Vibration

Brian E. Anderson, Cochair
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Scott D. Sommerfeldt, Cochair
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Invited Papers

1:00

2pSA1. Using sports equipment for student exploration of vibration and structural acoustics phenomena. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, drussell@engr.psu.edu)

The familiarity which many undergraduate and graduate students have with sports implements (bats, sticks, clubs, rackets, and balls), renders such objects as suitable subjects for student exploration through laboratory projects and homework problems. This paper will illustrate how problems and projects involving baseball bats, tennis rackets, hockey sticks, and golf clubs can arouse student interests in structural acoustics. Homework problems in which students model the contact time between ball and racket, the frequencies of a golf club shaft, the vibration of the face plate in a golf driver responsible for the trampoline effect, and the radiation of sound from structural modes in hollow and spherical balls provide engaging applications of the theory for membranes, flexural bending in beams, boundary conditions, and radiation. Examples will be shown for using baseball bats, cricket bats, hurling and hockey sticks, ping pong paddles, and tennis rackets in more involved laboratory projects. Sports implements also have the advantage of being non-uniform and asymmetric (wide body and thin handle), non-isotropic (wood grain), or involving complicated geometrics (ellipses) so that students can compare simple theory to more complicated realistic systems.

1:20

2pSA2. $e^{j\omega t}$ and the time domain in Introductory Engineering Acoustics. Karl Grosh (Dept. of Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109-2125, grosh@umich.edu)

Introductory classes in engineering acoustics traditionally focus on frequency domain solutions rather than those in the time domain. While this is somewhat historical in nature, it is also true that extensive coverage of analytic techniques for solving space-time partial differential equations is challenging in one semester. This is not a desirable situation because qualitative and quantitative understanding of the link between the frequency and time domains is often elusive to the new learner of acoustics. Such a deficit stands as a barrier to their deep understanding of the material as well as a knowledge gap of ubiquitous and important techniques. Naturally, computational tools to perform these manipulations are available along with high fidelity digital to analog conversion on students' personal computers. Students now have the capability to play back their simulations to their own ears to compare to experiment. I will present one approach to teaching these concepts to students with no assumed signal processing background using Fourier transforms. I will show how to weave a relatively complicated frequency domain simulation throughout the class and how this model problem can be demonstrated. This same problem also serves to introduce the effects of nonlinear distortion and clipping in transducers.

1:40

2pSA3. 3D-printing and vibration (including resonance). Lawrie Virgin (Dept. of Mech. Eng., Duke Univ., 141 Hudson Hall, Durham, NC 27708-0300, l.virgin@duke.edu)

The material contained in this paper focuses on using 3D printing of relatively simple, flexible structural components and plane frames. The relatively high resolution of modern 3D printers facilitates the production of slender structures, and thus provides an opportunity to exploit geometric parameter variations to enhance a practical understanding of stiffness and vibration. This approach has proved successful in initial inclusion in both the classroom via demonstration models, as well as in the lab in which elementary facilities can be utilized to acquire data. An especially useful facet of this approach is the assessment and justification for modeling simplifications, e.g., judging the circumstances under which the familiar sway assumption is valid in portal frames. Furthermore, minor parametric variations can be easily incorporated into sensitivity studies, and the production of multiple copies of essentially the same geometry promotes an effective use of statistical measurement uncertainty. We then extend this approach to forced vibration, with examples including the vibrating reed tachometer and sympathetic resonance.

2:00

2pSA4. The acoustic guitar as an experimental platform to teach vibration measurements, modal parameter estimation, and data management. Micah R. Shepherd and Tyler P. Dare (Appl. Res. Lab, Penn State Univ., PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu)

The laboratory class required by Penn State's Graduate Program in Acoustics teaches general experimental methods for acoustics. Students learn the basics of data acquisition, sensor calibration and excitation methods through lectures and performing experiments. This talk will discuss the lab experiment used to teach vibration measurements, basic modal parameter estimation and data management using an acoustic guitar as the experimental testbed. Students are required to develop their own test plan for accelerometer and impact hammer locations and then collect the data using a custom-written LabVIEW data acquisition program. Since the data acquisition involves multiple impact and response locations, the students are required to manage their data in a way that allows for easy mapping of data files to spatial locations. They then estimate the guitar body mode shapes by decomposing the measured mobility matrix into its singular values and right- and left-singular matrices. Finally, the students visualize their results by creating a mesh in MATLAB using the impact locations. By completing all of the steps to perform a modal analysis of an acoustic guitar, the students gain experience in both the practical considerations of modal analysis and experimentation in general.

2:20

2pSA5. Demonstrating structural waves in rods/beams. Scott D. Sommerfeldt (Dept of Phys., Brigham Young Univ., N249 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

Structures exhibit some interesting properties in that multiple types of waves can propagate, and the behavior of those waves is different. This demonstration focuses specifically on the difference between longitudinal waves in rods and bending waves in beams. As used here, rods and beams can be the same structure, in practice, but the terms are used to help distinguish between the particular type of wave propagation. For these wave types, the theory of wave propagation for rods and beams is quite accurate in predicting the experimental response, and can be readily demonstrated if near-ideal boundary conditions are used. This demonstration uses a rod/beam that has free-free boundary conditions. For longitudinal waves, the resonances of the rod should be perfectly harmonic. For bending waves, the resonances of the beam are unevenly spaced due to the dispersive nature of the bending waves. It will be shown how the results from a simple experimental apparatus match theory very well, and how the two types of structural waves are consistent with each other. This good agreement between theory and experiment can help reinforce these wave concepts in the minds of the students.

2:40

2pSA6. Case studies in teaching and research mentorship for undergraduates in structural acoustics and vibration. Donald B. Bliss, Mauricio Villa, David Raudales, and Linda P. Franzoni (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., 148B Hudson Hall, Durham, NC 27708, donald.bliss@duke.edu)

A hybrid educational model that mixes classroom learning and individual mentoring is presented for junior and senior undergraduates interested in structural acoustics and vibration. In this activity, talented undergraduates take Duke's first graduate level acoustics course and simultaneously take multiple semesters of independent study for academic credit. The independent studies are co-supervised by faculty members and graduate students. The student research projects are individualized and typically separate from ongoing graduate student research. This structure allows the undergraduate students to feel strong ownership, while giving graduate students teaching and mentoring experience, and providing faculty with a vehicle to explore new ideas. Projects typically have both analytical and experimental components that allow the students to develop their proficiency with visualization and computational tools while developing their physical insight. Several case studies are presented that show impressive results and substantial intellectual growth, including examples of long term commitment to a career in structural acoustics.

3:00 – 3:15

Contributed Papers

3:15

2pSA7. Seismic-infrasound-acoustic-meteorological sensors to dynamically monitor the natural frequencies of concrete dams. Henry Diaz-Alvarez, Luis A. De Jesus-Diaz, Vincent P. Chiarito, Christopher Simpson, and Mihan H. McKenna (GeoTech. Structures Lab., U.S. Army Engineer Res. and Development Ctr., 208 Gridners PL, Vicksburg, MS 39180, henry.diaz-alvarez@usace.army.mil)

The U.S. Army Engineer Research and Development Center (ERDC) is leading research using seismic-infrasound-acoustic-meteorological (SIAM) arrays to determine structural characteristics of critical infrastructure. Large infrastructure, such as dams, emit infrasound (acoustic energy below that of human perception) at their natural modes of vibration, which are related to their structural condition. To validate the concept and its potential use for monitoring flood control structures, a structural evaluation was conducted at the Portuguese Dam in Ponce, Puerto Rico. The dam's dynamic properties were studied prior to the deployment of SIAM arrays using detailed finite element (FE) models assembled in COMSOL Multiphysics software. The principal natural frequencies of the dam were identified and the fundamental modes were confirmed in the infrasound pass-band. The natural frequencies of 4.8 Hz, 6.7 Hz, and 10.2 Hz, respectively, were determined for the lower modes of vibrations. A total of three SIAM arrays were deployed multiple times over the course of three years. This provided a wide range of data to help understand how the dam's natural frequencies changed due to different loading conditions such as the different reservoir levels. Infrasound test results were compared to resonant frequencies calculated from the dam's transient responses produced by ambient and external load excitations.

3:30

2pSA8. Classroom demonstration of acoustic landmine detection using a clamped soil plate oscillator. Jenna M. Cartron (Phys. Dept., U.S. Naval Acad., 3519 Forest Haven Dr., Laurel, MD 20724, Jcartron11@gmail.com) and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD)

In landmine detection some mines (constructed out of plastic) are difficult to detect using ground penetrating radar. In acoustic landmine detection, one can detect a vibration profile across the surface of the soil (sand) due to the interaction between the soil column and the elastic top plate.

Table top classroom demonstrations can bring to life idealized field experiments. The soil plate oscillator (SPO) apparatus models an idealization of an acoustic landmine detection experiment. The clamped circular elastic plate models the top plate of a plastic buried mine while the soil column supported by the elastic plate models the burial depth. An amplified sinusoidal chirp from a sweep spectrum analyzer drives an AC coil placed below a small magnet attached to the underside of the clamped plate—causing vibration. A laser Doppler vibrometer measured the rms particle velocity at 13 scan locations across the 4.5 inch diameter soil surface in sweeps between 50 Hz and 850 Hz. Results show a fundamental mode shape at 184 Hz (with a central peak) and a second mode at 488 Hz (with peaks to the left and right of the null) corresponding to a second vibration mode.

3:45

2pSA9. Classroom demonstration of nonlinear tuning curve vibration and two-tone tests using a column of glass beads vibrating over a clamped elastic plate. Emily V. Santos and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, santosemily08@gmail.com)

A soil plate oscillator (SPO) apparatus consists of two circular flanges sandwiching and clamping a thin circular elastic plate. The apparatus can model the acoustic landmine detection problem. Here, uniform spherical glass beads—representing a nonlinear mesoscopic elastic material—are supported at the bottom by the acrylic plate (4.5 inch diam, 1/8 inch thick) and stiff cylindrical sidewalls of the upper flange. A magnetic disk centered and fastened below the plate is driven by an AC coil placed below the magnet. Nonlinear tuning curves of the magnet's acceleration are measured by driving the coil with a swept sinusoidal signal applied to a constant current amplifier. Ten (separate) tuning curve experiments are performed using a fixed column of 350 grams of beads using 1,2,3,…,10 mm diameter beads. The backbone curves (peak acceleration vs. corresponding resonant frequency) exhibit a linear region with comparable slopes, while the detailed curvature vs. bead diameter reveals more structure. A bilinear hysteresis model is useful for characterizing results. In two-tone tests, air-borne sound from two 3 inch diameter speakers drives the bead column surface at closely spaced frequencies near the fundamental resonance. Nonlinearly generated combination frequency tones are compared for each of the ten bead diameter experiments.

TUESDAY AFTERNOON, 8 MAY 2018 NICOLLET A, 1:00 P.M. TO 4:00 P.M.

Session 2pSC

Speech Communication: Clinical Populations (Poster Session)

Stephanie A. Borrie, Chair
Department of Communicative Disorders and Deaf Education, Utah State University, 1000 Old Main Hill, Logan, UT 84322

All posters will be on display from 1:00 p.m. to 4:00 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 1:00 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

2pSC1. The relationship between reading ability and categorical perception of spectrotemporal and spectral contrasts in children. Gabrielle O'Brien, Emily Kubota, and Jason D. Yeatman (Inst. for Learning and Brain Sci., Univ. of Washington, 1417 N.E. 42nd St., Box 354875, Seattle, WA 98105-6246, andronovhopf@gmail.com)

Behavioral evidence suggests that dyslexic readers are impaired in processing rapid temporal changes such as formant transitions. It is unclear if this possible impairment is attributable to the dynamic nature of a formant transition, the brevity of the cue, or a general deficit in categorical decision making. We tested categorical perception of speech in 30 children between the ages of 8 and 12 with a range of reading abilities (dyslexic, below average and above average readers) on two stimulus continua: (1) a /ba/-/da/ continuum, in which a 100-ms formant transition provides a spectrotemporal cue, and (2) a /sa/-/sha/ continuum with a purely spectral contrastive cue also lasting 100 ms. Two test conditions probed the interaction of working memory and categorical perception: a single-interval condition and an ABX paradigm. Children with dyslexia showed shallower psychometric functions and more lapses at the continuum endpoints compared to controls. There was no effect of stimulus continuum or test condition. This suggests that dyslexic children are not specifically impaired in their perception of formant transitions. Rather, dyslexic children may be subtly challenged in auditory coding of, or decision making about, temporally brief cues that may be either spectral or spectrotemporal in nature.

2pSC2. Mandarin vowel and tone identification for Mandarin congenital amusics. Mingshuang Li (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, limingshuang@utexas.edu), Wei Tang (State Key Lab. of Cognit. Neurosci. and Learning & IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., Beijing, China), Chang Liu (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX), Yun Nan, wenjing wang, and qi dong (State Key Lab. of Cognit. Neurosci. and Learning & IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., Beijing, Beijing, China)

Recent studies found that the lexical tone processing in Mandarin Chinese was normal for amusics with musical perception disorders, but was degraded for a subgroup of amusics, named tone agnosics. The current study aimed to further explore whether the lexical deficits in tone agnosics were presented in phonetic processing, in particular, diphthong and triphthong. Vowel and tone identifications were examined among three groups of Mandarin-native listeners: tone agnosics (amusics with lexical tone deficits), the pure amusics (with normal lexical processing) and normal controls. Preliminary results showed that tone agnosics had the lowest scores in the

identification of vowel-plus-tones, vowels, and tones, while the performances were similar between the pure amusics and normal controls. Moreover, tone agnosics showed deficits not only in lexical tone identification, but also in vowel identification. Particularly, the disadvantages of tone agnosics in both vowel and tone processing appeared to be greater in diphthongs and triphthongs than in monothongs. Speech processing of pure amusics was comparable with that of the control group, although they had deficits in musical disorders, implying that musical perception was separate from speech perception at high-level processing. On the other hand, tone agnosia had deficits in both musical and speech processing.

2pSC3. Effects of congenital blindness on the mismatch responses to Mandarin lexical tone, consonant, and vowel. Jie Feng, Xinchun Wu (Dept. of Psych., Beijing Normal Univ., No.19 XinJieKouWai St., Beijing 100875, China, myonly88@126.com), and Chang Liu (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX)

With limited visual input, blind individuals rely more on auditory information to perceive the environmental stimuli. In this study, we investigated the effect of congenital blindness on the Mismatch Negativity (MMN) to Mandarin lexical tones, consonants and vowels using the oddball paradigm. The study included 9 congenitally blind subjects and 10 normal-sighted subjects without hearing impairment, while they are matched on age and verbal IQ. For lexical tone processing, no significant difference was observed between the two groups in MMN latencies or amplitudes, while both groups had larger Mandarin tone MMN on the right and central than on the left. For consonant processing, the MMN amplitude was larger on the left and central than the right only for blind participants, whereas MMN latencies of consonants in both groups were shorter on the right than the left. For vowel processing, blind group had longer MMN latencies than control group. These results suggested that blind individuals had similar pre-attentive lexical tone processing, but they relied more on the left and central hemisphere to differentiate consonants, and it took them more time automatically to process different vowels than the sighted individuals, possibly resulting from the compensatory mechanism of neural reorganization in blind subjects.

2pSC4. Monitoring L-Dopa induced dyskinesias (LID) using speech acoustic measures in patients with Parkinson's disease. Emily Wang (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 600 South Paulina St., 1017 Armour, Chicago, IL 60612, emily_wang@rush.edu), Leonard A. Verhagen Metman (Neurological Sci., Rush Univ. Medical Ctr., Chicago, IL), Andong Zhan (Dept. of Comput. Sci., Johns Hopkins Univ., Baltimore, MD), and Carly Blodgett (RML Specialty Hospital in Chicago, Chicago, IL)

More than 90% of patients with Parkinson's disease (PD) treated with levodopa for over 5 years will develop motor fluctuations and involuntary movements known as levodopa-induced dyskinesias (LID). Since LID commonly manifested in the head-neck and trunk areas speech is inevitably affected. In this study, eight acoustic speech measures derived from three speech tasks were selected to examine changes in speech acoustics associated with LID. Twenty subjects with PD, ten during the medication-off state without dyskinesia, and ten during the medication-on state with dyskinesia completed the speech tasks. Results revealed that three speech acoustic measures demonstrated sensitivity to changes associated with LID. In order to capture the changes associated with LID throughout medication cycles, ten PD subjects were provided an Android phone with HopkinsPD, an Android phone application, installed. One-minute speech and video samples were simultaneously collected at three time points over a medication cycle (before, peak, and end dose), twice a day (first morning dose and first afternoon dose) and repeated over three days (every other day) at a patient's home. The results were discussed in light of the long-term goal to develop a reliable at home LID Smart-phone monitoring system that uses speech acoustic measures to monitor LID real-time remotely.

2pSC5. Vowel dynamics in persons with aphasia. Caroline A. Niziolek (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, 1500 Highland Dr., Rm. 485, Madison, WI 53705, cniziolek@wisc.edu) and Swathi Kiran (Speech, Lang. & Hearing Sci., Boston Univ., Boston, MA)

Although aphasia is primarily considered a disorder of language, higher-level grammatical and word-finding deficits are often accompanied by lower-level motoric deficits in the stable production of consonants and vowels. This project investigates the extent to which persons with aphasia (PWA) use feedback to monitor their speech by examining online formant movements during vowel production. PWA and age-matched controls produced 200 repetitions of three monosyllabic words ("eat", "Ed", "add") while neural activity was recorded using magnetoencephalography (MEG). This "speak" condition was interleaved with a "listen" condition in which recorded audio from the speak trials was played back to participants through earphones, serving as a baseline for the neural response. Vowel centering was defined as 2D formant (F1-F2) movement toward the vowel median over the time course of the syllable, lessening acoustic deviation. PWA had greater acoustic variability than controls at vowel onset, but both groups exhibited vowel centering, significantly decreasing variability over the course of the syllable. A hemispheric shift in neural responses to self-produced speech may accompany this corrective movement in PWA. These analyses inform theories of feedforward and feedback influences on speech production in aphasia.

2pSC6. Recovering prosody of a case of foreign accent syndrome. Grace Kuo (Foreign Lang. and Literatures, National Taiwan Univ., 1 Section 4, Roosevelt Rd., Taipei, Taipei City 106, Taiwan, graciakuo@ntu.edu.tw)

Foreign Accent Syndrome (FAS) is a rare disorder characterized by the emergence of a perceived foreign accent following brain damage. In this case study, acoustic analyses were performed on the speech of a Mandarin-speaking female FAS patient at her four doctor's appointments. The reading materials included news in newspaper and a tongue twister. The acoustic analyses include sentence-level intonation and rhythmic measures including %V and PVIs. Results reveal a gradual recovery trajectory from a disfluent stressed-timed pattern to a fluent syllable-timed pattern. A heritage Mandarin speaker and an advanced nonnative speaker recorded the same reading materials. Same acoustic analyses were performed on their speech for comparison.

2pSC7. What does it mean for a voice to be "normal?". Jody E. Kreiman (Head and Neck Surgery, UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu), Anita Auszmann (Linguist, UCLA, Los Angeles, CA), and Bruce R. Gerratt (Head and Neck Surgery, UCLA, Los Angeles, CA)

It is rather unclear what is meant by "normal" voice quality, just as it is unclear what is meant by "voice quality" in general. A clearer understanding of what listeners perceive as normal and what strikes them as disordered would benefit both clinical practice, for which a normal sound is presumably the goal of treatment, and the study of voice quality in general. To shed light on this matter, listeners heard 1-sec sustained vowels produced by 200 speakers (100 male and 100 female), half of whom were recorded in the clinic (ranging from mild to fairly severe pathology) and half of whom were UCLA students with no known vocal disorder. Listeners compared 20 voices at a time in a series of sort-and-rate trials, and ordered them in a line according to the severity of perceived vocal pathology. Any voices perceived as normal were placed in a box at one end of the line. Preliminary results indicate that listeners agreed fairly well about which voices were not normal, but not at all about which were normal. Implications of these findings for evaluation of voice in and out of the clinic will be discussed. [Work supported by NIH and NSF.]

2pSC8. The role of rhythm in adaptation to dysrhythmic speech, continued. Stephanie A. Borrie (Dept. of Communicative Disord. and Deaf Education, Utah State Univ., 1000 Old Main Hill, Logan, UT 84322, stephanie.borrie@usu.edu) and Kaitlin Lansford (School of Commun. Sci. and Disord., Florida State Univ., Tallahassee, FL)

Rhythm cues have been shown to be important for deciphering speech in adverse listening conditions, even when the rhythm cues are corrupted. In an initial attempt to document the relationship between rhythm perception and processing of a naturally dysrhythmic speech signal, we found that listeners with expertise in rhythm perception were not advantaged with initial intelligibility of ataxic dysarthria but were significantly advantaged with adaptation to the degraded speech signal [Borrie, Lansford, and Barrett, *Journal of Speech, Language, and Hearing Research,* **60**, 561–570 (2017)]. We speculated that listeners with skills in rhythm perception are better able to exploit experience (familiarization) with the degraded speech, learning

something useful about corrupted rhythm cues for subsequent processing. This current study investigated whether the relationship between rhythm perception and perceptual adaptation of dysrhythmic speech, observed in our previous study with ataxic dysarthria, holds for other forms of dysrhythmic speech. Here, we replicate our original study with two different forms of dysarthria: the largely irregular and unpredictable speech of *hyperkinetic* dysarthria secondary to Huntington's disease, and the relatively regular and predictable speech of *hypokinetic* dysarthric speech secondary to Parkinson's disease. Results will bear on models of dysrhythmic speech perception as well as clinical practice.

2pSC9. The impact of combined degradations: Intelligibility of disordered speech in noise. Sarah E. Yoho and Stephanie A. Borrie (Communicative Disord. and Deaf Education, Utah State Univ., 1000 Old Main Hill, Logan, UT 84321-6746, sarah.leopold@usu.edu)

In the current study, the intelligibility of dysarthric speech in background noise was determined. Speech-shaped noise was mixed with neurologically healthy (control) and disordered (dysarthric) speech at a series of signal-to-noise ratios. In addition, bandpass filtered control and dysarthric speech conditions were assessed to determine the effect of noise on both naturally and artificially degraded speech. Both the amount of noise and type of speech significantly impacted intelligibility, but there was no interaction between the two factors. Thus, it appears that there is no differential effect of noise on dysarthric speech relative to control speech. Despite this lack of interaction, it is important to note that the intelligibility of dysarthric speech was substantially lower than the intelligibility of control speech at each of the signal-to-noise ratios. This supports the idea that patients with dysarthria and their communication partners should be advised to select a favorable listening environment for communication. Lastly, large-scale online crowdsourcing via Amazon Mechanical Turk was utilized to collect data for the current study. Findings and implications for this data collection approach will be discussed.

2pSC10. Linguistic type-frequency in preschool boys and girls with and without hearing loss. Mark VanDam (Speech & Hearing Sci., Washington State Univ., PO Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu), Jenna Anderst (Commun. Sci. Disord., Eastern Washington Univ., Spokane, WA), Daniel Olds, Allison Saur (Speech & Hearing Sci., Washington State Univ., Spokane, WA), and Paul De Palma (Comput. Sci., Gonzaga Univ., Spokane, WA)

Linguistic type-frequency, how many different lexical types are used, has been examined in usage-based models of child language acquisition. In general, it has been shown that exposure to greater type-frequencies increases children's productive use of language and that language in turn bootstraps later development including language and literacy. It is not currently known if pediatric hearing loss impacts the type-frequency of those children's early communicative productions. In this study, we used a public database available via HomeBank [http://homebank.talkbank.org] to examine the type-frequency in 53 cognitively intact children, 37 with mild to moderate hearing loss (HL) and 16 peers who were typically-developing (TD). For each child, we analyzed 15 minutes of high volubility from a representative daylong recording collected in a natural family setting via an audio recorder worn by the child. Results indicate a main effect of sex favoring girls, but no main effect of HL. There were, however, interaction effects in which TD-boys had a greater type-frequency than HL-boys, but TD-girls were lesser than HL-girls. Within the HL group, HL-girls had greater type-frequency than HH-boys, but within the TD group TD-boys had greater type-frequency than TD-girls.

2pSC11. Audiovisual benefit in the perception of foreign-accented speech by cochlear-implant users. Emily Waddington, Brittany N. Jaekel, Anna R. Tinnemore, Sandra Gordon-Salant, and Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, 0100 Lefrak Hall, College Park, MD 20742, emily_waddington@yahoo.com)

When visual and auditory speech cues are presented together, listeners typically obtain an *audiovisual (AV) benefit*, or a boost in speech perception accuracy compared to auditory-only (AO) or visual-only (VO) presentations. Visual cues are particularly important for the speech perception of cochlear-implant (CI) users, who show a greater AV benefit than normal-hearing (NH) listeners. However, whether CI users can utilize visual cues for an AV benefit when the speaker has a foreign accent is unknown. Accented speech can alter auditory (primarily temporal) and visual cues, and is significantly more difficult for CI users in AO listening conditions. We investigated the AV benefit in CI users for unaccented and accented speech, presenting IEEE sentences in VO, AO, and AV conditions. The sentences were spoken by one native English speaker and one non-native English speaker who is a native Spanish-speaker from Peru. Preliminary results reveal that CI users show AV benefits in both unaccented and accented conditions, with substantially larger AV benefits for accented speech. Although visual cues alone may be insufficient for speech perception, CI users may necessitate more reliance on visual cues when presented with auditory cues that have been altered in accented speech.

2pSC12. Processing speed and age predict recognition of spectro-temporally degraded speech. Anna R. Tinnemore (Neurosci. and Cognit. Sci., Univ. of Maryland, College Park, 0100 Samuel J. LeFrak Hall, College Park, MD 20742, annat@terpmail.umd.edu), Lauren Evans (Hearing and Speech Sci., Univ. of Maryland, College Park, Silver Springs, MD), Sandra Gordon-Salant, and Matthew Goupell (Neurosci. and Cognit. Sci., Univ. of Maryland, College Park, College Park, MD)

Cochlear implants (CIs) provide a listener with the temporal envelope of an acoustic signal, but not the spectral fine structure. Speech recognition performance is highly variable among individuals and is often assessed clinically in ideal conditions (over-articulated, clean speech) rather than real-world listening environments. This study examined the effects of age and processing speed on understanding of time-compressed speech by listeners with normal hearing (NH) presented with vocoded speech of varying spectral resolution and listeners with CIs. NH listeners repeated sentences that had been time-compressed to varying degrees (no compression, 20%, 40%, or 60% time compression ratios) and then noise vocoded with various channels (4, 8, or 16 channels or unprocessed), while CI participants repeated unprocessed time-compressed sentences. Performance of age-matched NH listeners on the 4-channel vocoded speech most closely matched CI listeners' performance at each level of time-compression. For NH listeners, age effects were observed on conditions featuring time compression and reduced spectral resolution. Both NH and CI listener groups showed a significant interaction between age and time compression. As difficulty increased (greater time compression for all, fewer channels for NH), cognitive measures of processing speed were correlated with speech recognition performance across all age and listening groups.

2pSC13. How do differences in voice characteristics affect cochlear implant users' speech intelligibility in cocktail-party scenarios? Nawal El Boghdady (Dept. of Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen, Groningen, Groningen, Netherlands), Etienne Gaudrain (Cognition Auditive et Psychoacoustique, CNRS UMR 5292, Lyon Neurosci. Res. Ctr., Université Lyon 1, Lyon, FR, UMCG, KNO, Huispostcode BB20, PO box 30.001, Groningen 9700 RB, Netherlands, etienne.gaudrain@cnrs.fr), and Deniz Baskent (Dept. of Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen, Groningen, Netherlands)

Speech intelligibility in the presence of multiple competing talkers — the cocktail-party — is quite challenging for cochlear implant (CI) users. Previous literature has indicated abnormalities in CI users' perception of principal vocal cues (fundamental frequency, F0, and vocal-tract length, VTL), which have been shown to benefit normal hearing (NH) listeners in distinguishing between target and masking speakers in speech-on-speech (SOS) tasks. The hypothesis posited here is that CI users are also expected to benefit from voice differences between target and masking speakers, albeit at a smaller rate compared to NH listeners. Thus, in this study, the benefit in performance in two SOS tasks was measured for both NH (19 participants) and CI users (18 participants) as a function of systematically varying F0 and VTL of the masking talker relative to the target speaker. Results indicate that, contrary to the posited hypothesis and to NH listeners who benefit from differences in F0 and VTL between target and masker, CI users (i) do not benefit from differences in F0, and (ii) demonstrate a slight

decrease in performance as the difference in VTL is increased. These data have implications on sound coding strategies that should be developed to address these perceptual deficits in CI users. [**Funding:** The University Medical Center Groningen (UMCG), Advanced Bionics, Netherlands Organization for Scientific Research (NWO), VICI Grant No. 91817603.]

2pSC14. Developing finer-grained methods for analyzing word recognition performance by individuals with cochlear implants. Tzu-Ling J. Yu (Speech-Language-Hearing Sci., Univ. of Minnesota, Twin Cities, 1983 Knapp St., Apt. E4, Falcon Heights, MN 55108, yuxxx583@umn.edu), Sarah J. Rosen (Speech-Language-Hearing Sci., Univ. of Minnesota, Twin Cities, St. Paul, MN), Benjamin Munson, and Robert S. Schlauch (Speech-Language-Hearing Sci., Univ. of Minnesota, Twin Cities, Minneapolis, MN)

Cochlear implant (CI) listeners experience overall improvement in speech recognition abilities, but a considerable amount of variability still remains among these individuals. Data from simulated CI listening conditions also show wide variability in monosyllabic word recognition ability. In an adverse simulated condition (2 spectral channels using sinusoidal carriers), the word score can be nearly 0 for some adults with normal hearing. These floor effects pose a challenge for researchers, as they lead to data compression. When analyzing the performance by phonemes, the increase in scores may remove the floor effect. The purpose of this study is to establish the chance performance rate for NU-6 consonant-vowel-consonant (CVC) word lists analyzed by phonemes. To determine this, we randomly paired words from the NU-6 list with CVC words from an on-line corpus of English, the English Lexicon Project, and scored the matches by phonemes using dictionary pronunciations. Thousands of runs of these random pairings were used to generate expected baseline accuracy ranges. These results were compared with individual, behavioral data from people listening to NU-6 words processed using 2-channel sinusoidal vocoders to determine if participants were performing better than chance when scoring by phonemes.

2pSC15. Quality and quantity of infant-directed speech by maternal caregivers predicts later speech-language outcomes in children with cochlear implants. Laura Dilley, Elizabeth Wieland, Matthew Lehet, Meisam K. Arjmandi (Michigan State Univ., Dept. of Communicative Sci. and, East Lansing, MI 48824, ldilley@msu.edu), Derek Houston (The Ohio State Univ., Columbus, OH), and Tonya Bergeson (Butler Univ., Indianapolis, IN)

Caregivers speaking to children often adjust segmental and suprasegmental qualities of their speech relative to adult-directed (AD) speech. The quality and quantity of infant-directed (ID) speech has been shown to support word learning and word segmentation by normal-hearing infants, but the extent to which children with cochlear implants (CIs) benefit linguistically from ID speech is unclear. The present study investigated the extent to which the quantity and quality of ID speech produced in the lab by each of 40 mothers to her child with a CI predicted the child's speech-language outcome measures at two years post-implantation. Multiple measures of ID and AD speech for each mother were taken, including ID speech quantity in one minute, and several measures of ID speech quality, including fundamental frequency characteristics, speech rate, and the area of the vowel triangle formed by corner vowels in F1-F2 space. Forward stepwise regression showed that both quantity and quality of speech significantly predicted language outcomes measured by the Preschool Language Scales, Peabody Picture Vocabulary Test, and the Reynell Developmental Language Scales. These results support the hypothesis that hearing more ID speech that has acoustic modifications typical of IDS promotes language proficiency in children with CIs.

TUESDAY AFTERNOON, 8 MAY 2018 — GREENWAY H/I, 1:00 P.M. TO 5:15 P.M.

Session 2pSP

Signal Processing in Acoustics, Psychological and Physiological Acoustics, Architectural Acoustics, and Noise: Acoustics of Virtual and Augmented Reality

Buye Xu, Cochair
Signal Processing Research, Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344

Jens Meyer, Cochair
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Invited Papers

1:00

2pSP1. Robust data-independent object-related transfer function-based beamformer design for remote listening applications. Hendrik Barfuss, Michael Buerger, and Walter Kellermann (Chair of Multimedia Communications and Signal Processing, Friedrich-Alexander-Universität Erlangen-Nürnberg, Wetterkreuz 15, Erlangen 91058, Germany, hendrik.barfuss@FAU.de)

Virtual reality offers users to experience remote acoustic scenes which are captured in the original environment by a microphone array carried by, e.g., a third person or a robot. To extract and preserve the spatiotemporal nature of specific sounds of interest in the original acoustic environments, data-independent acoustic beamforming appears to be a suitable technique. In this work, we present recent work on robust data-independent beamformer design which is applicable to unconstrained sensor topologies and allows for an intuitive control of the beamformer's robustness. Combined with the concept of polynomial beamforming, flexible beam steering in real

time is possible. Consequently, the beamformer design is well suited for remote listening applications. If the beamformer design is applied to a microphone array, which is integrated into a scatterer, object-related transfer functions (ORTFs) need to be incorporated into the beamformer design. We briefly discuss selected approaches to obtaining the required ORTFs in this work. Finally, we present exemplary results of our robust beamformer design, carried out for a 12-element microphone array integrated into the head of a humanoid robot.

1:20

2pSP2. A planar array of differential microphones for 3D sound capture. Thushara D. Abhayapala, Prasanga N. Samarasinghe, and Hanchi Chen (Res. School of Eng., The Australian National Univ., Canberra, ACT 2601, Australia, thushara.abhayapala@anu.edu.au)

Three dimensional soundfield decomposition based on spherical harmonic analysis is becoming an integral part of 3D audio for virtual and augmented reality systems. Spherical harmonic analysis reveals the underlying characteristics of the soundfield, thus allowing high accuracy manipulation and analysis of the soundfield. This requires a microphone array with 3D pick up capability to detect the soundfield. A well-studied type of such array configuration is the spherical array. Since its geometry coincides with the spherical harmonics, the sound signal captured by a spherical microphone array is well-suited for the spherical harmonic transform. The placement of microphones on a spherical array has to follow a strict rule of orthogonality of the spherical harmonics which limits the flexibility of the array configuration. The spherical shape of the array also pose difficulties on implementation as well as practical usage. Recently, the authors presented the theory and design of a compact hybrid microphone array distributed on a 2D plane for three dimensional soundfield analysis. The proposed array uses a combination of omnidirectional microphones and differential microphones placed along multiple circles on the X-Y plane. In this paper, we present the practical implementation of a prototype array based on the above design.

1:40

2pSP3. Modal beamformer analysis of ambisonics decoders. Jens Meyer (mh Acoust., 38 Meade Rd., Fairfax, VT 05454, jmm@mhacoustics.com), Gary W. Elko (mh Acoust., Summit, NJ), and Steven Backer (mh Acoust., San Francisco, CA)

Ambisonic recording and playback is experiencing a revival with the introduction of higher-order Ambisonics (HOA) and is included in the recent MPEG-H and ATSC 3.0 standards. Higher-Order Ambisonics processing results in an improved spatial resolution compared to the original first-order Ambisonics proposed in 1970s. From a signal processing point of view, Ambisonics is a two-step process. In the first step spherical harmonic signals are generated by an Ambisonic encoder. Based on these spherical harmonic signals an Ambisonic decoder generates the output signals in the second step. Many different decoder flavors have emerged over time. Various decoder designs are motivated by different optimization criterion for the resulting soundfield. In this presentation we will analyze the Ambisonic decoders based on the fact that the two-step Ambisonics processing is closely related to spherical harmonic modal beamforming. It is shown that the decoder output signals can be interpreted in terms of beampatterns. The resulting beampatterns for the different decoders are computed and some performance criteria are presented.

2:00

2pSP4. Directional emphasis in ambisonics. W. Bastiaan Kleijn (Google, Inc., Wellington, New Zealand), Andrew Allen, Jan Skoglund, and Felicia S. Lim (Google, Inc., 345 Spear St., San Francisco, CA, bitllama@google.com)

We describe an ambisonics enhancement method that increases the signal strength in specified directions at a low computational cost. The strengthening method is referred to as a utilization of an enhancement operator. The operator can be used in a static setup to emphasize the signal arriving from a particular direction or set of directions, much like a spotlight amplifies the visibility of objects in one direction. It can also be used in an adaptive arrangement where it sharpens the directionality and reduces the distortions in timbre associated with low-degree ambisonics representations. The enhancement operator can be applied directly to time-domain ambisonics representations but also to time-frequency ambisonics representations.

2:20

2pSP5. A hybrid real-time auralization system for binaural reproduction of virtual acoustic environments with simulated room acoustics adapted for hearing aid users. Florian Pausch, Lukas Aspöck (ITA, RWTH Aachen Univ., Aachen, NRW, Germany), Michael Vorlaender, and Janina Fels (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

A recently developed binaural real-time reproduction system has been extended with an interface to research hearing aids allowing for the conduction of auditory research on subjects with hearing loss. Simulated hearing aid signals based on measurements of generic hearing aid-related transfer functions are additionally processed on a master hearing aid to emulate conventional hearing aid algorithms and played back through the hearing aid receivers. Designed for subjects with mild to moderate hearing loss, the system also facilitates the use of residual hearing capabilities by simulating an external sound field based on generic head-related transfer functions which is reproduced via loudspeakers and acoustic crosstalk cancellation filters. For increased ecological validity, plausible room acoustics are simulated using adjusted simulation models relying on geometrical acoustics. The proposed system was evaluated objectively on different levels by investigating the listening environment and various system components, running a benchmark analysis on the acoustical simulation and auralization, and measuring the combined system latency.

2:40–2:55 Break

2:55

2pSP6. High-resolution room virtualization using first-order source-and-receiver-directional room impulse responses. Franz Zotter, Franck Zagala, Markus Zaunschirm, and Matthias Frank (Inst. of Electron. Music and Acoust., Univ. of Music and Performing Arts, Inffeldfasse 10/3, Graz 8010, Austria, zotter@iem.at)

Virtual acoustic environments benefit from involving the source and receiver directivity to become fully interactive. In this contribution we discuss virtualization of a room based on measured source-and-receiver-directional room impulse responses (SRD RIR), captured by first-order-directional source and receiver arrays. Low order arrays may in some ways overcome a directional versus temporal resolution trade off one encounters in directional measurements. This is done by employing the spatial decomposition method of room impulse responses on both the first-order source and receiver side. We discuss the result of these resolution-enhanced and efficiently measured SRD RIR based on a comparative listening experiment involving the variable-directivity icosahedral loudspeaker (IKO), a spherical beamforming loudspeaker array, whose perceptual sculptural effects have been studied in a room. For the virtualization of the IKO, high resolution directivity measurement of the IKO are inserted, and for its rendering, a high-resolution set of head related impulse responses (HRIRs).

3:15

2pSP7. A raytracing method to create inhomogeneous wave fields for collaborative virtual reality systems. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

A new rapid, raytracing method was developed and implemented for collaborative virtual reality systems. The goal of the method was to auralize historic acoustic venues in Rensselaer's Collaborative-Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab). The CRAIVE-Lab provides a physical/digital environment for collaborative tasks using seamless video from multiple projectors and a 128-channel wave-field system with 6 additional ceiling loudspeakers. For the raytracing method, scanned floor plans are marked up by the user to create a two-dimensional geometric room model to calculate the temporal and spatial early-reflection pattern for the horizontal array of 128 loudspeakers resulting in an individual impulse response for each of the 128 loudspeakers. For this purpose, the loudspeaker array is virtually placed within the floor plan to calculate the impulse responses at each loudspeaker position considering the angles of incidence. The user can define wall materials and a directivity pattern for each sound source. The late reverberation tail is generated using a stochastic model based on the volume and surface characteristics of the space. An individual exponentially decaying reverberation tail is computed for each of the 134 loudspeakers with frequency-specific decay times. [Work supported by NSF Grant Nos. #1229391 and CISL.]

3:35

2pSP8. Achieving realism and repeatability of an orchestra simulated within a concert hall. Matthew T. Neal and Michelle C. Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mtn5048@psu.edu)

Virtual acoustic techniques can allow for direct comparison between concert halls, but care must be taken to ensure the virtual reconstruction connects back to the subtle perceptions of the real environment. To achieve better realism with repeatability between measured halls, a compact spherical loudspeaker array is being used to simulate an orchestra. This sound source is placed at twenty different source locations around the stage. For each instrument measurement position, a swept-sine signal is processed to radiate from the source with a particular instrument's directivity, and the spatial room impulse response (RIR) is measured. This processing is accomplished through filters designed from an instrument directivity database measured with 32 microphones. For each instrument, a set of full-frequency filters, one for each loudspeaker driver, were designed from one-third octave band spherical harmonic weights. These processing techniques are directly analogous to higher order Ambisonics (HOA). In the audience, a 32-element spherical microphone array is used to capture the spatial RIR, and each measurement is convolved with single-instrument anechoic recordings and auralized using HOA. Separate auralizations of each instrument are combined together into a full-orchestral auralization. The measurement setup, filter design, and equalization techniques will be presented. [Work supported by NSF Award 1302741.]

3:55

2pSP9. Perceived stability and localizability in binaural reproduction with sparse HRTF measurement grids. Zamir Ben-Hur (Oculus & Facebook and Ben-Gurion Univ. of the Negev, Be'er Sheva, Be'er Sheva 8410501, Israel, zami@post.bgu.ac.il), David L. Alon (Oculus & Facebook, Ashkelon, Israel), Boaz Rafaely (Elec. & Comput. Eng., Ben-Gurion Univ. of the Negev, Beer Sheva, Israel), and Ravish Mehra (Oculus & Facebook, Redmond, Washington)

With the increased popularity of virtual reality applications, the need for high fidelity spatial audio has emerged. Reproduction of high quality spatial audio requires high resolution individualized head-related transfer functions (HRTFs). However, these are typically unavailable as they demand a large number of measurements and specialized equipment. Given sparse measurements, it is necessary to spatially interpolate the HRTF before employing it in binaural reproduction. Prior studies suggested the use of spherical-harmonics (SH) representation as the basis for the interpolation. However, interpolation of sparse measurements may lead to errors due to spatial aliasing and SH series truncation. In this study, the effects of these errors on the perceived acoustic scene stability and on the localizability of the virtual sound source are investigated numerically, as well as perceptually. Experimental results indicate a significant effect on both attributes due to truncation error, while the effect of the aliasing error is less dominant.

4:15

2pSP10. A computational framework for objective assessment of spatial audio wavefields. Mark F. Bocko and Steven Crawford (ECE Dept., Univ. of Rochester, Rochester, NY 14627, bocko@ece.rochester.edu)

Fusion of binaural acoustic stimuli by humans to form spatial perceptions of virtual sound sources may be described in terms of the cross-coherence properties of a binaurally sampled acoustic wavefield. In this paper, a computational framework for predicting listener perception of acoustic wavefields is presented. A plane-wave expansion of the wavefield, with component amplitudes and phases described by random variables, provides a complete description of the wavefield's spatio-temporal coherence properties. The acoustic source locations and the signals driving the sources, the listening space acoustic response, the location, head orientation, and physiological characteristics of the listener's pinnae, head, and shoulders, and an explicit model of the human auditory system, including the peripheral auditory system and central binaural fusion process, comprise the full model. The source signals serve as inputs to the model and listener percepts indicating the locations and extent in space of the perceived virtual acoustic sources, are the outputs. The framework, which can accommodate constituent sub-models with varying levels of detail and sophistication, provides a useful framework to support the ongoing development of spatial sound rendering systems for applications such as cinema, teleconferencing, and virtual and augmented reality.

4:30

2pSP11. A customizable artificial auditory fovea. Christopher N. Casebeer and Ross K. Snider (Elec. and Comput. Eng., Montana State Univ., Cobleigh 541, Bozeman, MT 59715, christopher.casebee1@msu.montana.edu)

Neuroscience shows that auditory neurons extract a variety of specific signal parameters such as frequency, amplitude modulation, frequency modulation, and onsets. Furthermore, biology gives us examples of auditory foveae in bats and owls where there is overrepresentation of behaviorally relevant signal parameters. This shows that biology finds nuances of these parameters to be more important than compact coding considerations such as orthonormal basis functions. In contrast to typical spectrogram approaches where a specific time window is chosen and where the Fast Fourier Transform provides non-overlapping orthogonal rectangular tilings, we start with the smallest tiling possible. The uncertainty principle gives us this tiling as the Gaussian modulated sinusoid. We can rotate this tiling in the time-frequency plane, which gives us a chirplet. We then construct many chirplets (a chirplet set) centered at the same time-frequency point. We then find the chirplet that best matches the signal at this point. Our ability to sample at arbitrary locations in the time-frequency plane with different chirplet sets gives us the ability to create customizable artificial auditory foveae. We use this chirplet front-end for a classification task of identifying Marmoset vocalizations and comparisons to typical spectrogram methods for audio classification are made.

4:45

2pSP12. Automatic recognition and immersive representation of environmental soundscapes. Mallory M. Morgan and Jonas Braasch (Rensselaer Polytechnic Inst., Greene Bldg, RPI, Troy, NY 12180, morgam11@rpi.edu)

The goal of this research is to automatically analyze a spatial auditory scene using a 16-channel spherical microphone array and produce an automatic transcript of events that occurred in different directions. The system can be used to establish an automatic protocol for collaborative business meetings or to monitor natural environments. This paper focuses on the automatic data collection of environmental and animal sounds in conjunction with the RPI/IBM Jefferson Project to sensor Lake George, NY. The amount of audio data required for bioacoustics monitoring and other applications is often too large to manually sort the data. Consequently, automatic environmental sound recognition techniques have to be applied to automatically (1) identify and extract relevant acoustic stimuli and (2) classify these stimuli after a training period. Using spatial audio data collected over days in forested areas in Upstate New York, information is automatically extracted and analyzed to find the most relevant parts of these large data sets. The analysis can then be used to create an automated non-linear time-lapse to summarize the events that occurred and meaningfully re-represent them in our immersive virtual environment, CRAIVE-Lab —together with the collected visual material. [Work supported by NSF No. 1631674 and CISL.]

5:00

2pSP13. Multiple sound-source localization and identification with a spherical microphone array and lavalier microphone data. Jonathan Mathews and Jonas Braasch (Architectural Sci. - Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St, Greene Bldg., Troy, NY 12180, mathej4@rpi.edu)

Spherical microphone arrays have attained a considerable amount of interest for their dynamic beamforming and tracking capabilities. Although many robust algorithms have been developed to accurately localize individual sources, multiple source localization, especially for concurrent or colocated sources, has proved to be a challenging topic. This project demonstrates a technique to isolate individual concurrent sources by using lavalier microphone data to determine the relative difference in signal energy between sources. This data is then incorporated into the source-tracking algorithm governing the beamforming spherical array, a particle filter, by allowing the rejection of frames containing concurrent source information. This improves the tracking ability of the system. [Work supported by Rensselaer HASS Fellowship, NSF No. 1631674, and CISL.]

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 Drive 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 that those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 8 May 2018.

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.

TUESDAY AFTERNOON, 8 MAY 2018 MINNEHAHA, 3:15 A.M. TO 4:30 P.M.

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

P. B. Nelson, Vice Chair ASC S3
Department of SLHS, University of Minnesota, 115 Shevlin, 164 Pilsbury Drive S.E., Minneapolis, MN 55455

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 that those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 8 May 2018.

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, 8 MAY 2018 MINNEHAHA, 4:45 A.M. TO 5:45 P.M.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

R. J. Peppin, Chair ASC S1
5012 Macon Road, 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 Road
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. 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 that those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 8 May 2018.

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.

TUESDAY EVENING, 8 MAY 2018 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 hold its meeting 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.)	Greenway D
Acoustical Oceanography	Greenway A
Animal Bioacoustics	Lakeshore B
Architectural Acoustics	Nicollet D3
Musical Acoustics	Greenway F/G
Physical Acoustics	Greenway J
Psychological and Physiological Acoustics	Greenway D2
Structural Acoustics and Vibration	Greenway C