Session 2aAA

Architectural Acoustics and Engineering Acoustics: Architectural Acoustics and Audio I

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

Alexander U. Case, Cochair
Sound Recording Technology, University of Massachusetts Lowell, 35 Wilder St., Suite 3, Lowell, MA 01854

Chair's Introduction—7:55

Invited Papers

8:00

2aAA1. Excessive reverberance in an outdoor amphitheater. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The historic Ford Theatre in Hollywood, CA, is undergoing an overall renovation and expansion. Its centerpiece is the inexplicably asymmetrical 1200 seat outdoor amphitheater, built of concrete in 1931 after the original 1920 wood structure was destroyed by a brush fire in 1929, and well before the adjacent Hollywood Freeway was nearly as noisy as now. Renovation includes reorienting seating for better symmetry while maintaining the historic concrete, and improving audio, lighting, and support spaces. Sited within an arroyo overlooking a busy highway, and in view of the Hollywood Bowl, the new design features an expanded "sound wall" that will help to mitigate highway noise while providing optimal lighting and control positions. New sound-absorptive treatments will address the Ford's excessive reverberation, currently more than might be anticipated for an entirely outdoor space. The remarkably uniform distribution of ambient noise and apparent contributions by the arroyo to the reverberation will be discussed, along with assorted design challenges.

8:20

2aAA2. Room acoustics analysis, recordings of real and simulated performances, and integration of an acoustic shell mock up with performers for evaluation of a choir shell design. David S. Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

The current renovation of the 1883 Galloway Memorial Methodist Church required the repair and replacement of a number of room finishes, as well as resolution of acoustic problems related to their choir loft. This paper will present the various approaches used to determine the best course of action using primarily an *in-situ* analysis that includes construction mockups, simulated sources, and critical listening.

8:40

2aAA3. A decade later: What we've learned from The Pritzker Pavilion at Millennium Park. Jonathan Laney, Greg Miller, Scott Pfeiffer, and Carl Giegold (Threshold Acoust., 53 W Jackson Blvd., Ste. 815, Chicago, IL 60604, jlaney@thresholdacoustics.com)

Each design and construction process yields a building and systems that respond to a particular client at a particular time. We launch these projects into the wild and all too frequently know little of their daily lives and annual cycles after that. Occasionally, though, we have the opportunity to stay close enough to watch a project wear in, weather (sometimes literally), and respond to changing client and patron dynamics over time. Such is the case with the reinforcement and enhancement systems at the Pritzker Pavilion in Chicago's Millennium Park. On a fine-grained scale, each outdoor loudspeaker is individually inspected for its condition at the end of each season. Signal-processing and amplification equipment is evaluated as well, so the overall system is maintained at a high degree of readiness and reliability. Strengths and weaknesses of these components thereby reveal themselves over time. We will discuss these technical aspects as well as changing audience behaviors, modifications made for special events, and the ways all of these factors inform the future of audio (and video) in the Park.

9:00

2aAA4. An electro-acoustic conundrum—Improving the listening experience at the Park Avenue Armory. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com) and Paul Scarbrough (Akustiks, South Norwalk, CT)

Larger than a hanger for a commercial airliner, the Park Avenue Armory occupies an entire city block in midtown Manhattan. Its massive internal volume generates reverberation time in excess of three seconds. However, it functions as a true multi-purpose venue with programming that includes dramatic performances produced by the Manchester International Festival, and musical performances sponsored by Lincoln Center. We will discuss the unique nature of the venue as well as the tools and techniques employed in staging different productions.

9:20

2aAA5. Sound reinforcement in an acoustically challenging multipurpose space. Deb Britton (K2 Audio, 4900 Pearl East Circle, Ste. 201E, Boulder, CO 80301, deb@k2audio.com)

Often times, sound system designers are dealt less than ideal cards: design a sound reinforcement system that will provide great speech intelligibility, in a highly reverberant space, without modifying any of the architectural finishes. While this can certainly be a challenge, add to those prerequisites, the additional complication of the sound system serving a multi-purpose use, where different types of presentations must take place in different locations in the space, and with varying audience sizes. This paper presents a case study of such a scenario, and describes the approach taken in order to achieve the client's goals.

9:40

2aAA6. Comparison of source stimulus input method on measured speech transmission index values of sound reinforcement systems. Neil T. Shade (Acoust. Design Collaborative, Ltd., 7509 Lhirondelle Club Rd., Ruxton, MD 21204, nts@akustx.com)

One purpose of a sound reinforcement system is to increase the talker's speech intelligibility. A common metric for speech intelligibility evaluation is the Speech Transmission Index (STI) defined by IEC-60268-16 Revision 4. The STI of a sound reinforcement system can be measured by inputting a stimulus signal into the sound system, which is modified by the system electronics, and radiated by the sound system loudspeakers to the audience seats. The stimulus signal can be input via a line level connection to the sound system or by playing the stimulus signal through a small loudspeaker that is picked-up by a sound system microphone. This latter approach factors the entire sound system signal chain from microphone input to loudspeaker output. STI measurements were performed on two sound systems, one in a reverberant room and the other in relatively non-reverberant room. Measurement results compare both signal input techniques using omnidirectional and hypercardioid sound system microphones and three loudspeakers claimed to be designed to have directivity characteristics similar to the human voice.

10:00–10:20 Break

10:20

2aAA7. Enhancements in technology for improving access to active acoustic solutions in multipurpose venues. Ronald Freiheit (Wenger Corp., 555 Park Dr., Owatonna, MN 55060, ron.freiheit@wengercorp.com)

With advancements in digital signal processing technology and higher integration of functionality, access to active acoustics systems for multipurpose venues has been enhanced. One of the challenges with active acoustics systems in multipurpose venues is having enough control over the various acoustic environments within the same room (e.g., under balcony versus over balcony). Each area may require its own signal processing and control to be effective. Increasing the signal processing capacity to address these different environments will provide a more effective integration of the system in the room. A new signal processing platform with the flexibility to meet these needs is discussed. The new platform addresses multiple areas with concurrent processing and is integrated with a digital audio buss and a network-based control system. The system is flexible in its ability to easily expand to meet the needs of a variety of environments. Enhancing integration and flexibility of scale accelerates the potential for active systems with an attractive financial point of entry.

10:40

2aAA8. Sound levels and the risk of hearing damage at a large music college. Thomas J. Plsek (Brass, Berklee College of Music, MS 1140 Brass, 1140 Boylston St., Boston, MA 02215, tplsek@berklee.edu)

For a recent sabbatical from Berklee College of Music, my project was to study hearing loss especially among student and faculty musicians and to measure sound levels in various performance situation ranging from rehearsals to classes/labs to actual public performances. The National Institute for Occupational Safety and Health (NIOSH) recommendations (85 dBA criterion with a 3 dB exchange rate) were used to determine the daily noise dose obtained in each of the situations. In about half of the situations 100% or more of the daily noise was reached. More measuring of actual levels reached is needed as are noise dosimetry measurements over an active 12–16 hour day.

11:00

2aAA9. Development of a tunable absorber/diffuser using micro-perforated panels. Matthew S. Hildebrand (Wenger Corp., 555 Park Dr., Owatonna, MN 55060, matt.hildebrand@wengercorp.com)

Shared rehearsal spaces are an all-too-common compromise in music education, pitting vocal, and instrumental ensembles against each other for desirable room acoustics. More than ever, adjustable acoustics are needed in music spaces. An innovative new acoustic panel system was developed with this need for flexibility in mind. Providing variable sound absorption with a truly static aesthetic,

control of reverberation time in the mid-frequency bands is ultimately handed over to the end user. New product development test methods and critical design decisions are discussed, such as curving the micro-perforated panel to improve scattering properties. *In situ* reverberation measurements are also offered against a 3D CAD model prediction using lab-tested material properties.

11:20

2aAA10. Real case measurements of inflatable membranes absorption technique. Niels W. Adelman-Larsen (Flex Acoust., Diplomvej 377, Kgs. Lyngby 2800, Denmark, nwl@flexac.com)

After some years of development of the patented technology of inflated plastic membranes for sound absorption, an actual product became available in 2012 and immediately implemented in a Danish music school. It absorbs sound somewhat linearly from 63 to 1k Hz, when active, advantageous for amplified music. The absorption coefficient is close to 0.0 when deactivated. 75.000 ft^2 of the mobile version of the innovation was employed at the Eurovision Song Contest, the second largest annual television event worldwide. This contributed to a lowering of T30 in the 63, 125, and 250 Hz octave bands from up to 13 s to below 4 s in the former-shipyard venue. The permanently installed version has been incorporated in a new theater in Korea. More detailed acoustic measurements from these cases will be presented. The technology will further be used in the new, multi-functional Dubai Opera scheduled for 2015.

11:40

2aAA11. Virtual sound images and virtual sound absorbers misinterpreted as supernatural objects. Steven J. Waller (Rock Art Acoust., 5415 Lake Murray Blvd. #8, La Mesa, CA 91942, wallersj@yahoo.com)

Complex sound behaviors such as echoes, reverberation, and interference patterns can be mathematically modeled using the modern concepts of virtual sound sources or virtual sound absorbers. Yet prior to the scientific wave theory of sound, these same acoustical phenomena were considered baffling, and hence led to the illusion that they were due to mysterious invisible sources. Vivid descriptions of the physical forms of echo spirits, hoofed thunder gods, and pipers' stones, as engendered from the sounds they either produced or blocked, are found in ancient myths and legends from around the world. Additional pieces of evidence attesting to these beliefs are found in archaeological remains consisting of canyon petroglyphs, cave paintings, and megalithic stone circles. Blindfolded participants in acoustic experimental set-ups demonstrated that they attributed various virtual sound effects to real sound sources and/or attenuators. Ways in which these types of sonic phenomena can be manipulated to give rise to ultra-realistic auditory illusions of actual objects even today will be discussed relative to enhancing experiences of multimedia entertainment and virtual reality. Conversely, understanding how the mind can construct psychoacoustic models inconsistent with scientific reality could serve as a lesson helping prevent the supernatural misperceptions to which our ancestors were susceptible.

TUESDAY MORNING, 28 OCTOBER 2014 — LINCOLN, 8:25 A.M. TO 12:00 NOON

Session 2aAB

Animal Bioacoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Mobile Autonomous Platforms for Bioacoustic Sensing

Holger Klinck, Cochair
Cooperative Institute for Marine Resources Studies, Oregon State University, Hatfield Marine Science Center, 2030 SE Marine Science Drive, Newport, OR 97365

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

Chair's Introduction—8:25

Invited Papers

8:30

2aAB1. Real-time passive acoustic monitoring of baleen whales from autonomous platforms. Mark F. Baumgartner (Biology Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS #33, Woods Hole, MA 02543, mbaumgartner@whoi.edu)

An automated low-frequency detection and classification system (LFDCS) was developed for use with the digital acoustic monitoring (DMON) instrument to detect, classify, and report in near real time the calls of several baleen whale species, including fin, humpback, sei, bowhead, and North Atlantic right whales. The DMON/LFDCS has been integrated into the Slocum glider and APEX profiling float, and integration projects are currently underway for the Liquid Robotics wave glider and a moored buoy. In a recent

evaluation study, two gliders reported over 25,000 acoustic detections attributed to fin, humpback, sei, and North Atlantic right whales over a 3-week period during late fall in the Gulf of Maine. The overall false detection rate for individual calls was 14%, and for right, humpback, and fin whales, false predictions of occurrence during 15-minute reporting periods were 5% or less. Agreement between acoustic detections and visual sightings from concurrent aerial and shipboard surveys was excellent (9 of 10 visual detections were accompanied by real-time acoustic detections of the same species by a nearby glider). We envision that this autonomous acoustic monitoring system will be a useful tool for both marine mammal research and mitigation applications.

8:50

2aAB2. Detection, bearing estimation, and telemetry of North Atlantic right whale vocalizations using a wave glider autonomous vehicle. Harold A. Cheyne (Lab of Ornithology, Cornell Univ., 95 Brown Rd., Rm. 201, Ithaca, NY 14850, haroldcheyne@gmail.com), Charles R. Key, and Michael J. Satter (Leidos, Long Beach, MS)

Assessing and mitigating the effects of anthropogenic noise on marine mammals is limited by the typically employed technologies of archival underwater acoustic recorders and towed hydrophone arrays. Data from archival recorders are analyzed months after the activity of interest, so assessment occurs long after the events and mitigation of those activities is impossible. Towed hydrophone arrays suffer from nearby ship and seismic air gun noise, and they require substantial on-board human and computing resources. This work has developed an acoustic data acquisition, processing, and transmission system for use on a Wave Glider, to overcome these limitations by providing near real-time marine mammal acoustic data from a portable and persistent autonomous platform. Sea tests have demonstrated the proof-of-concept with the system recording four channels of acoustic data and transmitting portions of those data via satellite. The system integrates a detection-classification algorithm on-board, and a beam-forming algorithm in the shore-side user interface, to provide a user with aural and visual review tools for the detected sounds. Results from a two-week deployment in Cape Cod Bay will be presented and future development directions will be discussed.

9:10

2aAB3. Shelf-scale mapping of fish sound production with ocean gliders. David Mann (Loggerhead Instruments Inc., 6576 Palmer Park Circle, Sarasota, FL 34238, dmann@loggerhead.com), Carrie Wall (Univ. of Colorado at Boulder, Boulder, CO), Chad Lembke, Michael Lindemuth (College of Marine Sci., Univ. of South Florida, St.. Petersburg, FL), Ruoying He (Dept Marine, Earth, and Atmospheric Sci., NC State Univ., Raleigh, NC), Chris Taylor, and Todd Kellison (Beaufort Lab., NOAA Fisheries, Beaufort, NC)

Ocean gliders are a powerful platform for collecting large-scale data on the distribution of sound-producing animals while also collecting environmental data that may influence their distribution. Since 2009, we have performed extensive mapping on the West Florida Shelf with ocean gliders equipped with passive acoustic recorders. These missions have revealed the distribution of red grouper as well as identified several unknown sounds likely produced by fishes. In March 2014, we ran a mission along the shelf edge from Cape Canaveral, FL to North Carolina to map fish sound production. The Gulf Stream and its strong currents necessitated a team effort with ocean modeling to guide the glider successfully to two marine protected areas. This mission also revealed large distributions of unknown sounds, especially on the shallower portions of the shelf. Gliders provide valuable spatial coverage, but because they are moving and most fish have strong diurnal sound production patterns, data analysis on presence and absence must be made carefully. In many of these cases, it is best to use a combination of platforms, including fixed recorders and ocean profilers to measure temporal patterns of sound production.

9:30

2aAB4. The use of passively drifting acoustic recorders for bioacoustic sensing. Jay Barlow, Emily Griffiths, and Shannon Rankin (Marine Mammal and Turtle Div., NOAA-SWFSC, 8901 La Jolla Shores Dr., La Jolla, CA 92037, jay.barlow@noaa.gov)

Passively drifting recording systems offer several advantages over autonomous underwater or surface vessels for mobile bioacoustic sensing in the sea. Because they lack of any propulsion, self noise is minimized. Also, vertical hydrophone arrays are easy to implement, which is useful in estimating the distance to specific sound sources. We have developed an inexpensive (<$5000) Drifting Acoustic Spar Buoy Recorder (DASBR) that features up to 1 TB of stereo recording capacity and a bandwidth of 10 Hz–96 kHz. Given their low cost, many more recorders can be deployed to achieve greater coverage. The audio and GPS recording system floats at the surface, and the two hydrophones (at 100 m) are de-coupled from wave action by a dampner disk and an elastic cord. During a test deployment in the Catalina Basin (Nov 2013) we collected approximately 1200 hours of recordings using 5 DASBRs recording at 192 kHz sampling rate. Each recorder was recovered (using GPS and VHF locators) and re-deployed 3–4 times. Dolphin whistles and echo-location clicks were detectable approximately half of the total recording time. Cuvier's beaked whales were also detected on three occasions. Cetacean density estimation and ocean noise measurements are just two of many potential uses for free-drifting recorders.

9:50

2aAB5. Small cetacean monitoring from surface and underwater autonomous vehicles. Douglas M. Gillespie, Mark Johnson (Sea Mammal Res. Unit, Univ. of St. Andrews, Gatty Marine Lab., St Andrews, Fife KY16 8LB, United Kingdom, dg50@st-andrews.ac.uk), Danielle Harris (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, Fife, United Kingdom), and Kalliopi Gkikopoulou (Sea Mammal Res. Unit, Univ. of St. Andrews, St. Andrews, United Kingdom)

We present results of Passive Acoustic surveys conducted from three types of autonomous marine vehicles, two submarine gliders and a surface wave powered vehicle. Submarine vehicles have the advantage of operating at depth, which has the potential to increase detection rate for some species. However, surface vehicles equipped with solar panels have the capacity to carry a greater payload and currently allow for more on board processing which is of particular importance for high frequency odontocete species. Surface vehicles are also more suited to operation in shallow or coastal waters. We describe the hardware and software packages developed for each vehicle type and give examples of the types of data retrieved both through real time telemetry and recovered post deployment. High frequency echolocation clicks and whistles have been successfully detected from all vehicles. Noise levels varied considerably between vehicle types, though all were subject to a degree of mechanical noise from the vehicle itself.

10:10–10:35 Break

10:35

2aAB6. A commercially available sound acquisition and processing board for autonomous passive acoustic monitoring platforms.
Haru Matsumoto, Holger Klinck, David K. Mellinger (CIMRS, Oregon State Univ., 2115 SE OSU Dr., Newport, OR 97365, haru.matsumoto@oregonstate.edu), and Chris Jones (Embedded Ocean Systems, Seattle, WA)

The U.S. Navy is required to monitor marine mammal populations in U.S. waters to comply with regulations issued by federal agencies. Oregon State University and Embedded Ocean Systems (EOS) co-developed a passive acoustic data acquisition and processing board called Wideband Intelligent Signal Processor and Recorder (WISPR). This low-power, small-footprint system is suitable for autonomous platforms with limited battery and space capacity, including underwater gliders and profiler floats. It includes a high-performance digital signal processor (DSP) running the uClinux operating system, providing extensive flexibility for users to configure or reprogram the system's operation. With multiple WISPR-equipped mobile platforms strategically deployed in an area of interest, operators on land or at sea can now receive information in near-real time about the presence of protected species in the survey area. In April 2014, WISPR became commercially available via EOS. We are implementing WISPR in the Seaglider and will conduct a first evaluation test off the coast of Oregon in September. System performance, including system noise interference, flow noise, power consumption, and file compression rates in the data-logging system, will be discussed. [Funding from the US Navy's Living Marine Resources Program.]

10:55

2aAB7. Glider-based passive acoustic marine mammal detection. John Hildebrand, Gerald L. D'Spain, and Sean M. Wiggins (Scripps Inst. of Oceanogr., UCSD, Mail Code 0205, La Jolla, CA 92093, jhildebrand@ucsd.edu)

Passive acoustic detection of delphinid sounds using the Wave Glider (WG) autonomous near-surface vehicle was compared with a fixed bottom-mounted autonomous broadband system, the High-frequency Acoustic Recording Package (HARP). A group of whistling and clicking delphinids was tracked using an array of bottom-mounted HARPs, providing ground-truth for detections from the WG. Whistles in the 5–20 kHz band were readily detected by the bottom HARPs as the delphinids approached, but the WG revealed only a brief period with intense detections as the animals approached within ~500 m. Refraction due to acoustic propagation in the thermocline provides an explanation for why the WG may only detect whistling delphinids at close range relative to the long-range detection capabilities of the bottom-mounted HARPs. This work demonstrated that sound speed structure plays an important role in determining detection range for high-frequency-calling marine mammals by autonomous gliders and bottom-mounted sensors.

Contributed Papers

11:15

2aAB8. Acoustic seagliders for monitoring marine mammal populations. Lora J. Van Uffelen (Ocean and Resources Eng., Univ. of Hawaii at Manoa, 1000 Pope Rd., MSB 205, Honolulu, HI 96815, loravu@hawaii.edu), Erin Oleson (Cetacean Res. Program, NOAA Pacific Islands Fisheries Sci. Ctr., Honolulu, HI), Bruce Howe, and Ethan Roth (Ocean and Resources Eng., Univ. of Hawaii at Manoa, Honolulu, HI)

A DMON digital acoustic monitoring device has been integrated into a Seaglider with the goal of passive, persistent acoustic monitoring of cetacean populations. The system makes acoustic recordings as it travels in a sawtooth pattern between the surface and up to 1000 m depth. It includes three hydrophones, located in the center of the instrument and on each wing. An onboard real-time detector has been implemented to record continuously after ambient noise has risen above a signal-to-noise (SNR) threshold level, and the glider transmits power spectra of recorded data back to a shore-station computer via iridium satellite after each dive. The glider pilot has the opportunity to set parameters that govern the amount of data recorded, thus managing data storage and therefore the length of a mission. This system was deployed in the vicinity of the Hawaiian Islands to detect marine mammals as an alternative or complement to conventional ship-based survey methods. System design and implementation will be described and preliminary results will be presented.

11:30

2aAB9. Prototype of a linear array on an autonomous surface vehicle for the register of dolphin displacement patterns within a shallow bay.
Eduardo Romero-Vivas, Fernando D. Von Borstel-Luna (CIBNOR, Instituto Politécnico Nacional 195, Playa Palo de Santa Rita Sur, La Paz, BCS 23090, Mexico, evivas@cibnor.mx), Omar A. Bustamante, Sergio Beristain (Acoust. Lab, ESIME, IPN, IMA, Mexico City, Mexico), Miguel A. Porta-Gándara, Francisco Villa Médina, and Joaquín Gutiérrez-Jagüey (CIBNOR, La Paz, BCS, Mexico)

A semi-resident population of tursiops has been reported in the south of La Paz bay in Baja California Sur, Mexico, where specific zones for social, feeding and resting behaviors have been detected. Nevertheless, increasing human activities and new constructions are attributed to have shifted the areas of their main activity. Therefore, it becomes important to study displacement patterns of dolphins within the bay and their spatial relationship to maritime traffic and other sources of anthropogenic noise. A prototype of an Autonomous Surface Vehicle (ASV) designed for shallow water bathymetry has been adapted to carry a linear array of hydrophones previously reported for the localization of dolphins from their whistles. Conventional beam-forming algorithms and electrical steering are used to find Direction of Arrival (DOA) of the sound sources. The left-right ambiguity typical of a linear array and front-back lobes for sound sources located at end-fire can be resolved by the trajectory of the ASV. Geo-referenced positions and bearing of the array, provided by the Inertial Measurement Unit of the ASV, along with DOA for various positions allows triangulating and mapping the sound sources. Results from both, controlled experiments using geo-referenced know sources, and field trials within the bay, are presented.

11:45

2aAB10. High-frequency observations from mobile autonomous platforms. Holger Klinck, Haru Matsumoto, Selene Fregosi, and David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Hatfield Marine Sci. Ctr., 2030 SE Marine Sci. Dr., Newport, OR 97365, Holger.Klinck@oregonstate.edu)

With increased human use of US coastal waters—including use by renewable energy activities such as the deployment and operation of wind, wave, and tidal energy converters—the issue of potential negative impacts on coastal ecosystems arises. Monitoring these areas efficiently for marine mammals is challenging. Recreational and commercial activities (e.g., fishing) can hinder long-term operation of fixed moored instruments. Additionally these shallow waters are often utilized by high-frequency cetaceans (e.g., harbor porpoises) which can only be acoustically detected over short distances of a few hundred meters. Mobile acoustic platforms are a useful tool to survey these areas of concern with increased temporal and spatial resolution compared to fixed systems and towed arrays. A commercially available acoustic recorder (type Song Meter SM2+, Wildlife Acoustics, Inc.) featuring sampling rates up to 384 kHz was modified and implemented on an autonomous underwater vehicle (AUV) as well as an unmanned surface vehicle (USV) and tested in the field. Preliminary results indicate that these systems are effective at detecting the presence of high-frequency cetaceans such as harbor porpoises. Potential applications, limitations, and future directions of this technology will be discussed. [Project partly supported by ONR and NOAA.]

TUESDAY MORNING, 28 OCTOBER 2014 INDIANA G, 8:25 A.M. TO 12:00 NOON

Session 2aAO

Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Parameter Estimation in Environments That Include Out-of-Plane Propagation Effects

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

Chair's Introduction—8:25

Invited Papers

8:30

2aAO1. Estimating waveguide parameters using horizontal and vertical arrays in the vicinity of horizontal Lloyd's mirror in shallow water. Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

When shallow water internal waves approach a source-receiver track, the interference between the direct and horizontally refracted acoustic paths from a broadband acoustic source was previously shown to form Horizontal Lloyd's mirror (Badiey *et al*. J. Acoust. Soc. Am. 128(4), EL141–EL147, 2011). While the modal interference structure in the vertical plane may reveal arrival time for the out of plane refracted acoustic wave front, analysis of moving interference pattern along the horizontal array allows measurement of the angle of horizontal refraction and the speed of the nonlinear internal wave (NIW) in the horizontal plane. In this paper we present a full account of the movement of NIW towards a source-receive track and how we can use the received acoustic signal on an L-shaped array to estimate basic parameters of the waveguide and obtain related temporal and spatial coherence functions particularly in the vicinity of the formation of the horizontal Lloyd mirror. Numerical results using Vertical Modes and Horizontal Rays as well as 3D PE calculations are carried out to explain the experimental observations. [Work supported by ONR 322OA.]

8:50

2aAO2. Slope inversion in a single-receiver context for three-dimensional wedge-like environments. Frédéric Sturm (LMFA (UMR 5509 ECL-UCBL1-INSA de Lyon), Ecole Centrale de Lyon, Ctr. Acoustique, Ecole Centrale de Lyon, 36, Ave. Guy de Collongue, Ecully 69134, France, frederic.sturm@ec-lyon.fr) and Julien Bonnel (Lab-STICC (UMR CNRS 6285), ENSTA Bretagne, Brest Cedex 09, France)

In a single-receiver context, time-frequency (TF) analysis can be used to analyze modal dispersion of low-frequency broadband sound pulses in shallow-water oceanic environments. In a previous work, TF analysis was used to study the propagation of low-frequency broadband pulses in three-dimensional (3-D) shallow-water wedge waveguides. Of particular interest is that TF analysis turns out to be a suitable tool to better understand, illustrate and visualize 3-D propagation effects for such wedge-like environments. In the present work, it is shown that TF analysis can also be used at the core of an inversion scheme to estimate the slope of the seabed in a same single hydrophone receiving configuration and for similar 3-D wedge-shaped waveguides. The inversion algorithm proposed, based on a masking process, focuses on specific parts of the TF domain where modal energy is concentrated. The criterion used to quantify the match between the received signal and the replicas by a fully 3-D parabolic equation code, is defined as the amount of measured time-frequency energy integrated inside the masks. Its maximization is obtained using an exhaustive search. The method is first benchmarked on numerical simulations and then successfully applied on experimental small-scale data.

9:10

2aAO3. Effects of environmental uncertainty on source range estimates from horizontal multipath. Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

A method has been developed to estimate source range in continental shelf environments that exhibit three-dimensional propagation effects [M. S. Ballard, J. Acoust. Soc. Am. 134, EL340–EL343, 2013]. The technique exploits measurements recorded on a horizontal line array of a direct path arrival, which results from sound propagating across the shelf to the receiver array, and a refracted path arrival, which results from sound propagating obliquely upslope and refracting back downslope to the receiver array. A hybrid modeling approach using vertical modes and horizontal rays provides the ranging estimate. According to this approach, rays are traced in the horizontal plane with refraction determined by the modal phase speed. Invoking reciprocity, the rays originate from the center of the array and have launch angles equal to the estimated bearing angles of the direct and refracted paths. The location of the source in the horizontal plane is estimated from the point where the rays intersect. In this talk, the effects of unknown environmental parameters, including the sediment properties and the water-column sound-speed profile, on the source range estimate are discussed. Error resulting from uncertainty in the measured bathymetry and location of the receiver array will also be addressed. [Work supported by ONR.]

Contributed Papers

9:30

2aAO4. Acoustical observation of the estuarine salt wedge at low-to-mid-frequencies. D. Benjamin Reeder (Oceanogr., Naval Postgrad. School, 73 Hanapepe Loop, Honolulu, HI 96825, dbreeder@nps.edu)

The estuarine environment often hosts a salt wedge, the stratification of which is a function of the tide's range and speed of advance, river discharge volumetric flow rate and river mouth morphology. Competing effects of temperature and salinity on sound speed control the degree of acoustic refraction occurring along an acoustic path. A field experiment was carried out in the Columbia River to test the hypothesis that the estuarine salt wedge is acoustically observable in terms of low-to-mid-frequency acoustic propagation. Linear frequency-modulated (LFM) acoustic signals in the 500–2000 Hz band were collected during the advance and retreat of the salt wedge during May 27–28, 2013. Results demonstrate that the three-dimensional salt wedge front is the dominant physical feature controlling acoustic propagation in this environment: received signal energy is relatively stable under single-medium conditions before and after the passage of the salt wedge front, but suffers a 10–15 dB loss as well as increased variance during salt wedge front passage due to 3D refraction and scattering. Physical parameters (i.e., temperature, salinity, current, and turbulence) and acoustic propagation modeling corroborate and inform the acoustic observations.

9:45

2aAO5. A hybrid approach for estimating range-dependent properties of shallow water environments. Michael Taroudakis and Costas Smaragdakis (Mathematics and Appl. Mathematics & IACM, Univ. of Crete and FORTH, Knossou Ave., Heraklion 71409, Greece, taroud@math.uoc.gr)

H hybrid approach based on statistical signal characterization and a linear inversion scheme for the estimation of range dependent sound speed profiles of compact support in shallow water is presented. The approach is appropriate for ocean acoustic tomography when there is a single receiver available, as the first stage of the method is based on the statistical characterization of a single reception using wavelet transform to associate the signal with a set of parameters describing the statistical features of its wavelet sub-band coefficients. A non-linear optimization algorithm is then applied to associate these features with range-dependent sound speed profile in the water column. This inversion method is restricted to cases where the range dependency is of compact support. At the second stage a linear inversion scheme based on modal arrivals identification and a first order perturbation formula to associate sound speed differences with modal travel time perturbations is applied to fine tune the results obtained by the optimization scheme. A second restriction of this stage is that mode identification is necessary. If this assumption is fulfilled the whole scheme may be applied in ocean acoustic tomography for the retrieval of three-dimensional features, combining inversion results at various slices.

10:00–10:15 Break

Invited Papers

10:15

2aAO6. Three-dimensional acoustics in basin scale propagation. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com) and Richard L. Campbell (OASIS Inc., Seattle, U.S. Minor Outlying Islands)

Long-range, basin-scale acoustic propagation has long been considered deep water and well represented by the two-dimensional numerical solution (range/depth) of wave equation. Ocean acoustic tomography has even recently been demonstrated to be insensitive to the three-dimensional affects of refraction and diffraction (Dushaw, JASA 2014). For frequencies below 50 Hz, where volume attenuation is negligible, the approximation that all propagation of significance is in the plane begins to break down. When examining very long-range propagation in situations where the source/receiver are not specifically selected for open water paths, 3D effects can dominate. Seamounts and bathymetry rises cause both refraction away from the shallowing seafloor and diffraction behind sharp edges. In

this paper a set of recent observations, many from the International Monitoring System (IMS) of the United Nations Comprehensive Test Ban Treaty Organization (CTBTO) will be presented, demonstrating observations that are not well explained by Nx2D acoustic propagation. The Peregrine PE model, a recent recoding of RAM in C, has been extended to include 3D split-step Pade propagation and will be used to demonstrate how 3D acoustic propagation affects help explains some of the observations.

10:35

2aAO7. Sensitivity analysis of three-dimensional sound pressure fields in complex underwater environments. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

A sensitivity kernel for sound pressure variability due to variations of index of refraction is derived from a higher-order three-dimensional (3D) split-step parabolic-equation (PE) solution of the Helmholtz equation. In this study, the kernel is used to compute the acoustic sensitivity field between a source and a receiver in a 3D underwater environment, and to quantify how much of the medium change can cause significant consequence on received acoustic signals. Using the chain rule, the dynamics of sensitivity fields can be connected to the dynamics of ocean processes. This talk will present numerical examples of sound propagation in submarine canyons and continental slopes, where the ocean dynamics cause strong spatial and temporal variability in sound pressure. Using the sensitivity kernel technique, we can analyze the spatial distribution and the temporal evolution of the acoustic sensitivity fields in these geologically and topographically complex environments. The paper will also discuss other applications of this sound pressure sensitivity kernel, including uncertainty quantification of transmission loss prediction and adjoint models for 3D acoustic inversions. [Work supported by the ONR.]

10:55

2aAO8. Sensitivity analysis of the image source method to out-of-plane effects. Samuel Pinson (Laboratório de Vibrações e Acústica, Universidade Federal de Santa Catarina, LVA Dept de Engenharia Mecânica, UFSC, Bairro Trindade, Florianópolis, SC 88040-900, Brazil, samuelpinson@yahoo.fr) and Charles W. Holland (Penn State Univ., State College, PA)

In the context of seafloor characterization, the image source method is a technique to estimate the sediment sound-speed profile from broadband seafloor reflection data. Recently the method has been extended to treat non-parallel layering of the sediment stack. In using the method with measured data, the estimated sound-speed profiles are observed to exhibit fluctuations. These fluctuations may be partially due to violation of several assumptions: (1) the layer interfaces are smooth with respect to the wavelength and (2) out-of-plane effects are negligible. In order to better understand the impact these effects, the sensitivity of the image source method to roughness and out-of-plane effects are examined.

Contributed Papers

11:15

2aAO9. Results of matched-field inversion in a three-dimensional oceanic environment ignoring horizontal refraction. Frédéric Sturm (LMFA (UMR 5509 ECL-UCBL1-INSA de Lyon), Ecole Centrale de Lyon, Ctr. Acoustique, Ecole Centrale de Lyon, 36, Ave. Guy de Collongue, Ecully 69134, France, frederic.sturm@ec-lyon.fr) and Alexios Korakas (Lab-STICC (UMR6285), ENSTA Bretagne, Brest Cedex 09, France)

For some practical reasons, inverse problems in ocean acoustics are often based on 2-D modeling of sound propagation, hence ignoring 3-D propagation effects. However, the acoustic propagation in shallow-water environments, such as the continental shelf, may be strongly affected by 3-D effects, thus requiring 3-D modeling to be accounted for. In the present talk, the feasibility and the limits of an inversion in fully 3-D oceanic environments assuming 2-D propagation are investigated. A simple matched-field inversion procedure implemented in a Bayesian framework and based on the exhaustive search of the parameter space is used. The study is first carried out on a well-established wedge-like synthetic test case, which exhibits well-known 3-D effects. Both synthetic data and replica are generated using a parabolic-equation-based code. This approach highlights the relevance of using 2-D propagation models when inversions are performed at relatively short ranges from the source. On the other hand, important mismatch occurs when inverting at farther ranges, demonstrating that the use of fully 3-D forward models is required. Results of inversion on experimental small-scale data, based on a subspace approach as suggested by the preliminary study made on the synthetic test case, are presented.

11:30

2aAO10. Measurements of sea surface effects on the low-frequency acoustic propagation in shallow water. Altan Turgut, Marshall H. Orr (Acoust. Div., Naval Res. Lab, Acoust. Div., Code 7161, Washington, DC 20375, altan.turgut@nrl.navy.mil), and Jennifer L. Wylie (Fellowships Office, National Res. Council, Washington, DC)

In shallow water, spatial and temporal variability of the water column often restricts accurate estimations of bottom properties from low-frequency acoustic data, especially under highly active oceanographic conditions during the summer. These effects are reduced under winter conditions having a more uniform sound-speed profile. However, during the RAGS03 winter experiment, significant low-frequency (200–500 Hz) acoustic signal degradations have been observed on the New Jersey Shelf, especially in the presence of frequently occurring winter storms. Both in-plane and out-of-plane propagation effects were observed on three moored VLAs and one bottom-moored HLA. These effects were further analyzed using 3-D PE simulations with inputs from a 3-D time-evolving surface gravity wave model. It is shown that higher-order acoustic modes are highly scattered at high sea states and out-of-plane propagation effects become important when surface-wave fronts are parallel to the acoustic propagation track. In addition, 3-D propagation effects on the source localization and geoacoustic inversions are investigated using the VLA data with/without the presence of winter storms. [Work supported by ONR.]

11:45

2aAO11. Effects of sea surface roughness on the mid-frequency acoustic pulse decay in shallow water. Jennifer Wylie (National Res. Council, 6141 Edsall Rd., Apt. H, Alexandira, VA 22304, jennie.wylie@gmail.com) and Altan Turgut (Acoust. Div., Naval Res. Lab., Washington, DC)

Recent and ongoing efforts to characterize sea bed parameters from measured acoustic pulse decay have neglected the effects of sea surface roughness. In this paper, these effects are investigated using a rough surface version of RAMPE, RAMSURF, and random rough surface realizations, calculated from a 2D JONSWAP sea surface spectrum with directional spreading. Azimuthal dependence is investigated for sandy bottoms and found that the rate of pulse decay increases when the surface wave fronts are perpendicular to the path of acoustic propagation and higher significant wave height results in higher decay rates. Additionally, the effects from sea surface roughness are found to vary with different waveguide parameters including but not limited to sound speed profile, water depth, and seabed properties. Of particular interest are the combined effects of sea bed properties and rough sea surfaces. It is shown that when clay like sediments are present, higher-order modes are strongly attenuated and effects due to interaction with the rough sea surface are less pronounced. Finally, possible influences of sea-state and 3D out-of-plane propagation effects on the seabed characterization efforts will be discussed. [Work supported by ONR.]

TUESDAY MORNING, 28 OCTOBER 2014 — INDIANA A/B, 7:55 A.M. TO 12:10 P.M.

Session 2aBA

Biomedical Acoustics: Quantitative Ultrasound I

Michael Oelze, Cochair
UIUC, 405 N. Mathews, Urbana, IL 61801

Jonathan Mamou, Cochair
F. L. Lizzi Center for Biomedical Engineering, Riverside Research, 156 William St., 9th Floor, New York, NY 10038

Chair's Introduction—7:55

Invited Papers

8:00

2aBA1. Myocardial tissue characterization: Myofiber-induced ultrasonic anisotropy. James G. Miller (Phys., Washington U Saint Louis, Box 1105, 1 Brookings Dr., Saint Louis, MO 63130, james.g.miller@wustl.edu) and Mark R. Holland (Radiology and Imaging Sci., Indiana Univ. School of Medicine, Indianapolis, IN)

One goal of this invited presentation is illustrate the capabilities of quantitative ultrasonic imaging (tissue characterization) to determine local myofiber orientation using techniques applicable to clinical echocardiographic imaging. Investigations carried out in our laboratory in the late 1970s were perhaps the first reported studies of the impact on the ultrasonic attenuation of the angle between the incoming ultrasonic beam and the local myofiber orientation. In subsequent studies, we were able to show that the ultrasonic backscatter exhibits a maximum and the ultrasonic attenuation exhibits a minimum when the sound beam is perpendicular to myofibers, whereas the attenuation is maximum and the backscatter is minimum for parallel insonification. Results from our laboratory demonstrate three broad areas of potential contribution derived from quantitative ultrasonic imaging and tissue characterization: (1) improved diagnosis and patient management, such as monitoring alterations in regional myofiber alignment (for example, potentially in diseases such as hypertrophic cardiomyopathy), (2) improved echocardiographic imaging, such as reduced lateral wall dropout in short axis echocardiographic images, and (3) improved understanding of myocardial physiology, such as contributing to a better understanding of myocardial twist resulting from the layer-dependent helical configuration of cardiac myofibers. [NIH R21 HL106417.]

8:20

2aBA2. Quantitative ultrasound for diagnosing breast masses considering both diffuse and non-diffuse scatterers. James Zagzebski, Ivan Rosado-Mendez, Haidy Gerges-Naisef, and Timothy Hall (Medical Phys., Univ. of Wisconsin, 1111 Highland Ave., Rm. L1 1005, Madison, WI 53705, jazagzeb@wisc.edu)

Quantitative ultrasound augments conventional ultrasound information by providing parameters derived from scattering and attenuation properties of tissue. This presentation describes our work estimating attenuation (ATT) and backscatter coefficients (BSC), and computing effective scatterer sizes (ESD) to differentiate benign from malignant breast masses. Radio-frequency echo data are obtained from patients scheduled for biopsy of suspicious masses following an institutional IRB approved protocol. A Siemens S2000 equipped with a linear array and recently a volume scanner transducer is employed. Echo signal power spectra are computed from the tissue and from the same depth in a reference phantom having accurately measured acoustic properties. Ratios of the tissue-to-reference power spectra enable tissue ATT and BSC's to be estimated. ESD's are then computed by fitting BSC vs. frequency results to a size-dependent scattering model. A heterogeneity index HDI expresses variability of the ESD over the tumor area. In preliminary data from 35 patients, a Bayesian classifier incorporating ATT, ESD, and HDI successfully differentiated malignant masses from fibroadenomas. Future work focuses on analysis methods when diffuse scattering and stationary signal conditions, implicitly assumed in the power spectra calculations, are not present. This approach tests for signal coherence and generates new parameters that characterize these scattering conditions.

8:40

2aBA3. Quantitative ultrasound translates to human conditions. William O'Brien (Elec. and Comput. Eng., Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801, wdo@uiuc.edu)

Two QUS studies will be discussed that demonstrate significant potential for translation to human conditions. One of the studies deals with the early detection of spontaneous preterm birth (SPTB). In a cohort of 68 adult African American women, each agreed to undergo up to five transvaginal ultrasound examinations for cervical ultrasonic attenuation (at 5 MHz) and cervical length between 20 and 36 weeks gestation (GA). At 21 weeks GA, the women who delivered preterm had a lower mean attenuation (1.02 ± 0.16 dB/cm MHz) than the women delivering at term (1.39 ± 0.095 dB/cm MHz), p = 0.041. Cervical length at 21 weeks was not significantly different between groups. Attenuation risk of SPTB (1.2 dB/cm MHz threshold at 21 weeks): specificity = 83.3%, sensitivity = 65.4%. The other QUS study deals with the early detection of nonalcoholic fatty liver disease (NAFLD). Liver attenuation (ATN) and backscattered coefficients (BSC) were assessed at 3 MHz and compared to the liver MR-derived fat fraction (FF) in a cohort of 106 adult subjects. At a 5% FF (for NAFLD, FF $\geq$ 5%), an ATN threshold of 0.78 dB/cm MHz provided a sensitivity of 89%, and specificity of 84%, whereas a BSC threshold of 0.0028/cm-sr provided a sensitivity of 92% and specificity of 96%.

9:00

2aBA4. Quantitative-ultrasound detection of cancer in human lymph nodes based on support vector machines. Jonathan Mamou, Daniel Rohrbach (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@rri-usa.org), Alain Coron (Laboratoire d'Imagerie Biomédicale, Sorbonne Universités and UPMC Univ Paris 06 and CNRS and INSERM, Paris, France), Emi Saegusa-Beecroft (Dept. of Surgery, Univ. of Hawaii and Kuakini Medical Ctr., Honolulu, HI), Thanh Minh Bui (Laboratoire d'Imagerie Biomédicale, Sorbonne Universités and UPMC Univ Paris 06 and CNRS and INSERM, Paris, France), Michael L. Oelze (BioAcoust. Res. Lab., Univ. of Illinois, Urbana-Champaign, IL), Eugene Yanagihara (Dept. of Surgery, Univ. of Hawaii and Kuakini Medical Ctr., Honolulu, HI), Lori Bridal (Laboratoire d'Imagerie Biomédicale, Sorbonne Universités and UPMC Univ Paris 06 and CNRS and INSERM, Paris, France), Tadashi Yamaguchi (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba, Japan), Junji Machi (Dept. of Surgery, Univ. of Hawaii and Kuakini Medical Ctr., Honolulu, HI), and Ernest J. Feleppa (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY)

Histological assessment of lymph nodes excised from cancer patients suffers from an unsatisfactory rate of false-negative determinations. We are evaluating high-frequency quantitative ultrasound (QUS) to detect metastatic regions in lymph nodes freshly excised from cancer patients. Three-dimensional (3D) RF data were acquired from 289 lymph nodes of 82 colorectal-, 15 gastric-, and 70 breast-cancer patients with a custom scanner using a 26-MHz, single-element transducer. Following data acquisition, individual nodes underwent step-sectioning at 50-μm to assure that no clinically significant cancer foci were missed. RF datasets were analyzed using 3D regions-of-interest that were processed to yield 13 QUS estimates including spectral-based and envelope-statistics-based parameters. QUS estimates are associated with tissue microstructure and are hypothesized to provide contrast between non-cancerous and cancerous regions. Leave-one-out classifications, ROC curves, and areas under the ROC (AUC) were used to compare the performance of support vector machines (SVMs) and step-wise linear discriminant analyses (LDA). Results showed that SVM performance (AUC = 0.87) was superior to LDA performance (AUC = 0.78). These results suggest that QUS methods may provide an effective tool to guide pathologists towards suspicious regions and also indicate that classification accuracy can be improved using sophisticated and robust classification tools. [Supported in part by NIH grant CA100183.]

9:20

2aBA5. Quantitative ultrasound assessment of tumor responses to chemotherapy using a time-integrated multi-parameter approach. Hadi Tadayyon, Ali Sadeghi-Naini, Lakshmanan Sannachi, and Gregory Czarnota (Dept. of Medical Biophys., Univ. of Toronto, 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, gregory.czarnota@sunnybrook.ca)

Radiofrequency ultrasound data were collected from 60 breast cancer patients prior to treatment and at during the onset of their several-month treatment, using a clinical ultrasound scanner operating a ~7 MHz linear array probe. ACE, SAS, spectral, and BSC parameters were computed from 2×2 mm RF segments within the tumor region of interest (ROI) and averaged over all segments to obtain a mean value for the ROI. The results were separated into two groups—responders and non-responders—based on the ultimate clinical/pathologic response based on residual tumor size and tumor cellularity. Using a single parameter approach, the best prediction of response was achieved using the ACE parameter (76% accuracy at week 1). In general, more favorable classifications were achieved using spectral parameter combinations (82% accuracy at week 8), compared to BSC parameter combinations (73% accuracy). Using the multi-parameter approach, the best prediction was achieved using the set [MBF, SS, SAS, ACE] and by combining week 1 QUS data with week 4 QUS data to predict the response at week 4, providing accuracy as high as 91%. The proposed QUS method may potentially provide early response information and guide cancer therapies on an individual patient basis.

9:40

2aBA6. Quantitative ultrasound methods for uterine-cervical assessment. Timothy J. Hall (Medical Phys., Univ. of Wisconsin, 1005 WIMR, 1111 Highland Ave., Madison, WI 53705, tjhall@wisc.edu), Helen Feltovich (Medical Phys., Univ. of Wisconsin, Park City, Utah), Lindsey C. Carlson, Quinton Guerrero, Ivan M. Rosado-Mendez, and Bin Huang (Medical Phys., Univ. of Wisconsin, Madison, WI)

The cervix is a remarkable organ. One of its tasks is to remain firm and “closed” (5 mm diameter cervical canal) prior to pregnancy. Shortly after conception the cervix begins to soften through collagen remodeling and increased hydration. As the fetus reaches full-term there is a profound breakdown in the collagen structure. At the end of this process, the cervix is as soft as warm butter and the cervical canal has dilated to about 100 mm diameter. Errors in timing of this process are a cause for preterm birth, which has a cascade of life-threatening consequences. Quantitative ultrasound is well-suited to monitoring these changes. We have demonstrated the ability to accurately assess the elastic properties and acoustic scattering properties (anisotropy in backscatter and attenuation) of the cervix in non-pregnant hysterectomy specimens and in third trimester pregnancy. We’ve shown that acoustic and mechanical properties vary along the length of the cervix. When anisotropy and spatially variability are accounted for, there are clear differences in parameter values with subtle differences in softening. We are corroborating acoustic observations with nonlinear optical microscopy imaging for a reality check on underlying tissue structure. This presentation will provide an overview of this effort.

10:00–10:10 Break

10:10

2aBA7. Characterization of anisotropic media with shear waves. Matthew W. Urban, Sara Aristizabal, Bo Qiang, Carolina Amador (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu), John C. Brigham (Dept. of Civil and Environ. Eng., Dept. of BioEng., Univ. of Pittsburgh, Pittsburgh, PA), Randall R. Kinnick, Xiaoming Zhang, and James F. Greenleaf (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN)

In conventional shear wave elastography materials are assumed to be linear, elastic, homogeneous, and isotropic. These assumptions are important to account for in certain tissues because they are not always appropriate. Many tissues such as skeletal muscle, the kidney, and the myocardium are anisotropic. Shear waves can be used to investigate the directionally dependent mechanical properties of anisotropic media. To study these tissues in a systematic way and to account for the effects of the anisotropic architecture, laboratory-based phantoms are desirable. We will report on several phantom-based approaches for studying shear wave anisotropy, assuming that these materials are transversely isotropic. Phantoms with embedded fibers were used to mimic anisotropic tissues. Homogeneous phantoms were compressed to induce transverse isotropy according to the acoustoelastic phenomenon, which is related to nonlinear behavior of the materials. The fractional anisotropy of these phantoms was quantified to compare with measurements made in soft tissues. In addition, soft tissues are also viscoelastic, and we have developed a method to model viscoelastic transversely isotropic materials with the finite element method (FEM). The viscoelastic property estimation from phantom experiments and FEM simulations will also be discussed.

10:30

2aBA8. Applications of acoustic radiation force for quantitative elasticity evaluation of bladder, thyroid, and breast. Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 200 1st St. SW, Rochester, MN 55905, fatemi@mayo.edu)

Acoustic radiation force (ARF) provides a simple and yet non-invasive mechanism to induce a localized stress inside human body. The response to this excitation is used to estimate the mechanical properties of the targeted tissue *in vivo*. This talk covers an overview of three studies that use ARF for estimation of elastic properties of thyroid, breast, and the bladder in patients. The studies on thyroid and breast were aimed at differentiating between malignant and benign nodules. The study on the bladder was aimed at indirect evaluation of bladder compliance; hence, only a global measurement was needed. The study on breast showed that 16 out of 18 benign masses and 21 out of 25 malignant masses were correctly identified. The study on 9 thyroid patients with 7 benign and 2 malignant nodules showed all malignant nodules were correctly classified and only 2 of the 7 benign nodules were misclassified. The bladder compliance study revealed a high correlation between our method and independent clinical measurement of compliance (R-squared of 0.8–0.9). Further investigations on larger groups of patients are needed to fully evaluate the performances of the methods.

10:50

2aBA9. Multiband center-frequency estimation for robust speckle tracking applications. Emad S. Ebbini and Dalong Liu (Elec. and Comput. Eng., Univ. of Minnesota, 200 Union St. SE, Minneapolis, MN 55455, ebbin001@umn.edu)

Speckle tracking is widely used for the detection and estimation of minute tissue motion and deformation with applications in elastography, shear-wave imaging, thermography, etc. The center frequency of the echo data within the tracking window is an important parameter in the estimation of the tissue displacement. Local variations in this quantity due to echo mixtures (specular and speckle components) may produce a bias in the estimation of tissue displacement using correlation-based speckle tracking methods. We present a new algorithm for estimation and tacking of the center frequency variation in pulse-echo ultrasound as a quantitative tissue property and for robust speckle tracking applications. The algorithm employs multiband analysis in the determination of echo mixtures as a pre-processing step before the estimation of the center frequency map. This estimate, in turn, is used to improve the robustness of the displacement map produced by the correlation-based speckle tracking. The performance of the algorithm is demonstrated in two speckle tracking applications of interest in medical ultrasound: (1) ultrasound thermography and (2) vascular wall imaging.

11:10

2aBA10. Echo decorrelation imaging for quantification of tissue structural changes during ultrasound ablation. T. Douglas Mast, Tyler R. Fosnight, Fong Ming Hooi, Ryan D. Keil, Swetha Subramanian, Anna S. Nagle (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu), Marepalli B. Rao (Environ. Health, Univ. of Cincinnati, Cincinnati, OH), Yang Wang, Xiaoping Ren (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Syed A. Ahmad (Surgery, Univ. of Cincinnati, Cincinnati, OH), and Peter G. Barthe (Guided Therapy Systems/Ardent Sound, Mesa, AZ)

Echo decorrelation imaging is a pulse-echo method that maps millisecond-scale changes in backscattered ultrasound signals, potentially providing real-time feedback during thermal ablation treatments. Decorrelation between echo signals from sequential image frames is spatially mapped and temporally averaged, resulting in images of cumulative, heat-induced tissue changes. Theoretical analysis indicates that the mapped echo decorrelation parameter is equivalent to a spatial decoherence spectrum of the tissue reflectivity, and also provides a method to compensate decorrelation artifacts caused by tissue motion and electronic noise. Results are presented from experiments employing 64-element linear arrays that perform bulk thermal ablation, focal ablation, and pulse-echo imaging using the same piezoelectric elements, ensuring co-registration of ablation and image planes. Decorrelation maps are shown to correlate with ablated tissue histology, including vital staining to map heat-induced cell death, for both ex vivo ablation of bovine liver tissue and *in vivo* ablation of rabbit liver with VX2 carcinoma. Receiver operating characteristic curve analysis shows that echo decorrelation predicts local ablation with greater success than integrated backscatter imaging. Using artifact-compensated echo decorrelation maps, heating-induced decoherence of tissue scattering media is assessed for *ex vivo* and *in vivo* ultrasound ablation by unfocused and focused beams.

11:30

2aBA11. Quantitative ultrasound imaging to monitor *in vivo* high-intensity ultrasound treatment. Goutam Ghoshal (Res. and Development, Acoust. MedSystems Inc., 208 Burwash Ave., Savoy, IL 61874, ghoshal2@gmail.com), Jeremy P. Kemmerer, Chandra Karunakaran, Rami Abuhabshah, Rita J. Miller, and Michael L. Oelze (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

The success of any minimally invasive treatment procedure can be enhanced significantly if combined with a robust noninvasive quantitative imaging modality. Quantitative ultrasound (QUS) imaging has been widely investigated for monitoring various treatment responses such as chemotherapy and thermal therapy. Previously we have shown the feasibility of using spectral based quantitative ultrasound parameters to monitor high-intensity focused ultrasound (HIFU) treatment of in situ tumors [Ultrasonic Imaging, 2014]. In the present study we examined the use the various QUS parameters to monitor HIFU treatment of an *in vivo* mouse mammary adenocarcinoma model. Spectral parameters in terms of the backscatter coefficient, integrated backscattered energy, attenuation coefficient, and effective scatterer size and concentration were estimated from radiofrequency signals during the treatment. The characteristic of each parameter was compared to the temperature profile recorded by needle thermocouple inserted into the tumor a few millimeters away from the focal zone of the intersecting HIFU and the imaging transducer beams. The changes in the QUS parameters during the HIFU treatment followed similar trends observed in the temperature readings recorded from the thermocouple. These results suggest that QUS techniques have the potential to be used for non-invasive monitoring of HIFU exposure.

11:50

2aBA12. Rapid simulations of diagnostic ultrasound with multiple-zone receive beamforming. Pedro Nariyoshi and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48824, mcgough@egr.msu.edu)

Routines are under development in FOCUS, the “Fast Object-oriented C++ Ultrasound Simulator” (http://www.egr.msu.edu/~fultras-web), to accelerate B-mode image simulations by combining the fast nearfield method with time-space decomposition. The most recent addition to the FOCUS simulation model implements receive beamforming in multiple zones. To demonstrate the rapid convergence of these simulations in the nearfield region, simulations of a 192 element linear array with an electronically translated 64 element sub-aperture are evaluated for a transient excitation pulse with a center frequency of 3 MHz. The transducers in this simulated array are 5 mm high and 0.5133 mm wide with a 0.1 mm center to center spacing. The simulation is evaluated for a computer phantom with 100,000 scatterers. The same configuration is simulated in Field II (http://field-ii.dk), and the impulse response approach with a temporal sampling rate of 1 GHz is used as reference. Simulations are evaluated for the entire B-mode image simulated with each approach. The results show that, with sampling frequencies of 15 MHz and higher, FOCUS eliminates all of the numerical artifacts that appear in the nearfield region of the B-mode image, whereas Field II requires much higher temporal sampling frequencies to obtain similar results. [Supported in part by NIH Grant R01 EB012079.]

Session 2aED

Education in Acoustics: Undergraduate Research Exposition (Poster Session)

Uwe J. Hansen, Chair
Chemistry & Physics, Indiana State University, 64 Heritage Dr., Terre Haute, IN 47803-2374

All posters will be on display from 9:00 a.m. to 11:00 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:00 a.m.

Contributed Papers

2aED1. Prediction of pressure distribution between the vocal folds using Bernoulli's equation. Alexandra Maddox, Liran Oren, Sid Khosla, and Ephraim Gutmark (Univ. of Cincinnati, 3317 Bishop St., Apt. 312, Cincinnati, OH 45219, maddoxat@mail.uc.edu)

Determining the mechanisms of self-sustained oscillation of the vocal folds requires characterization of intraglottal aerodynamics. Since most of the intraglottal aerodynamics forces cannot be measured experimentally, most of the current understanding of vocal fold vibration mechanism is derived from analytical and computational models. Several of such studies have used the Bernoulli's equation in order to calculate the pressure distribution between the vibrating folds. In the current study, intraglottal pressure measurements are taken in a hemilarynx model and are compared with pressure values that are computed form the Bernoulli's equation. The hemilarynx model was made by removing one fold and having the remaining fold vibrating against a metal plate. The plate was equipped with two pressure ports located near the superior and inferior aspects of the fold. The results show that pressure calculated using Bernoulli's equation matched well with the measured pressure waveform during the glottal opening phase and dissociated during the closing phase.

2aED2. Effects of room acoustics on subjective workload assessment while performing dual tasks. Brenna N. Boyd, Zhao Peng, and Lily Wang (Eng., Univ. of Nebraska at Lincoln, 11708 s 28th St., Bellevue, NE 68123, bnboyd@unomaha.edu)

This investigation examines the subjective workload assessments of individuals using the NASA Task Load Index (TLX), as they performed speech comprehension tests under assorted room acoustic conditions. This study was motivated due to the increasing diversity in US classrooms. Both native and non-native English listeners participated, using speech comprehension test materials produced by native English speakers in the first phase and by native Mandarin Chinese speakers in the second phase. The speech materials were disseminated in an immersive listening environment to each listener under 15 acoustic conditions, from combinations of background noise level (three levels from RC-30, 40, and 50) and reverberation time (five levels from 0.4 to 1.2 seconds). During each condition, participants completed assorted speech comprehension tasks while also tracing a moving dot for an adaptive rotor pursuit task. At the end of each acoustic condition, listeners were asked to assess the perceived workload by completing the six-item NASA TLX survey, e.g., mental demand, perceived performance, effort, and frustration. Results indicate that (1) listeners' workload assessments degraded as the acoustic conditions became more adverse, and (2) the decrement in subjective assessment was greater for non-native listeners.

2aED3. Analysis and virtual modification of the acoustics in the Nebraska Wesleyan University campus theatre auditorium. Laura C. Brill (Dept. of Phys., Nebraska Wesleyan Univ., 5000 St. Paul Ave, Lincoln, NE 68504, lbrill@nebrwesleyan.edu), Matthew G. Blevins, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Omaha, NE)

NWU's McDonald Theatre Auditorium is used for both musical and non-musical performances. The acoustics of the space were analyzed in order to determine whether the space could be modified to better fit its uses. The acoustic characteristics of the room were obtained from impulse responses using the methods established in ISO 3382-1 for measuring the acoustic parameters of a performance space. A total of 22 source/receiver pairs were used. The results indicate a need for increased reverberation in the mid to high frequency ranges of 500–8000 Hz. The experimental results were used to calibrate a virtual model of the space in ODEON acoustics software. Materials in the model were then successfully modified to increase reverberation time and eliminate unwanted flutter echoes to optimize the acoustics to better suit the intended purposes of the space.

2aED4. The diffraction pattern associated with the transverse cusp caustic. Carl Frederickson and Nicholas L. Frederickson (Phys. and Astronomy, Univ. of Central Arkansas, LSC 171, 201 Donaghey Ave., Conway, AR 72035, nicholaslfrederickson@gmail.com)

New software has been developed to evaluate the Pearcey function $P_{\pm}(w_1,w_2)=\int_{(-\infty)}^{\infty} \exp[\pm i(s^{4/4} + w_2 s^{2/2} + w_1 s)]ds$. This describes the diffraction pattern of a transverse cusp caustic. Run-time comparisons between different coding environments will be presented. The caustic surface produced by the reflection of a spherical wavefront from the surface given by $h(x,y) = h_{1x}^2 + h_{2xy}^2 + h_{3y}^2$ will also be displayed.

2aED5. Architectural acoustical oddities. Zev C. Woodstock and Caroline P. Lubert (Mathematics & Statistics, James Madison Univ., 301 Dixie Ave., Harrisonburg, VA 22801, lubertcp@jmu.edu)

The quad at James Madison University (Virginia, USA) exhibits an uncommon, but not unique, acoustical oddity called Repetition Pitch. When someone stands at certain places on the quad and makes a punctuated white noise (claps, for example) a most unusual squeak is returned. This phenomenon only occurs at these specific places. A similar effect has been observed in other locations, mostly notably Ursinus College (Pennsylvania, USA) and the pyramid at Chichen Itza (Mexico). This talk will discuss Repetition Pitch, as well as other interesting architectural acoustic phenomenon including the noisy animals in the caves at Arcy-sur-Cure (France), the early warning system at Golkonda Fort (Southern India) and the singing angels at Wells Cathedral in the United Kingdom.

2aED6. Impedance tube measurements of printed porous materials. Carl Frederickson and Forrest McDougal (Phys. and Astronomy, Univ. of Central Arkansas, LSC 171, 201 Donaghey Ave., Conway, AR 72035, FMCDOUGAL1@CUB.UCA.EDU)

An impedance tube has been used to make measurements of the acoustic impedance of porous samples. Porous with designed porosities and tortuosities have been produced using 3D printing. Measured impedances are compared to calculated values.

2aED7. Stick bombs: A study of the speed at which a woven stick construction self-destructs. Scotty McKay and William Slaton (Phys. & Astronomy, The Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72034, SMCKAY2@uca.edu)

A stick bomb is created by weaving sticks together in a particular pattern. By changing the way the sticks are woven together, different types of stick bombs are created. After the stick bomb is woven to the desired length, one side of the stick bomb can be released causing it to rapidly begin tearing itself apart in the form of a pulse that propagates down the weave. This occurs due to a large amount of potential energy stored within the multitude of bent sticks; however, the physics of this phenomena has not been studied to the authors knowledge. The linear mass density of the stick bomb can be changed by varying the tightness of the weave. Data on these stick bombs, including video analysis to determine the pulse speed, will be presented.

2aED8. Three-dimensional printed acoustic mufflers and aeroacoustic resonators. John Ferrier and William Slaton (Phys. & Astronomy, The Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72034, jpferrierjr@gmail.com)

We explore and present the use of 3D printing technology to design, construct, and test acoustic elements that could be used as a low-frequency Helmholtz-resonator style muffler in a ventilation duct system. Acoustic elements such as these could be quickly prototyped, printed, and tested for any noisy duct environment. These acoustic elements are tested with and without mean flow to characterize their sound absorption (and sound generation) properties. It is found that at particular ranges of air flow speeds the simply designed acoustic muffler acts as a site for aeroacoustic sound generation. Measurement data and 3D model files with Python-scripting will be presented for several muffler designs. This work is supported by the Arkansas Space Grant Consortium in collaboration with NASA's Acoustics Office at the Johnson Space Center.

2aED9. Determining elastic moduli of concrete using resonance. Gerard Munyazikwiye and William Slaton (Phys. & Astronomy, The Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72034, GMUNYAZIKWIYE1@uca.edu)

The elastic moduli of rods of material can be determined by resonance techniques. The torsional, longitudinal, and transverse resonance modes for a rod of known mass and length can be measured experimentally. These resonance frequencies are related to the elastic properties of the material, hence, by measuring these quantities the strength of the material can be determined. Preliminary tests as proof of principle are conducted with metallic rods. Data and experimental techniques for determining the elastic moduli for concrete using this procedure will be presented.

2aED10. Articulation of sibilant fricatives in Colombian Spanish. Alexandra Abell and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 404 West Kirkwood Ave., Bloomington, IN 47404, alabell@indiana.edu)

Colombians constitute the largest South American population in the United States at 909,000 (or 24% of the total South American population in the U.S.), and Bogotá, Colombia is the most populated area within the Andean Highland region, yet relatively little is known about Colombian Spanish speech production. The majority of previous studies of Colombian phonetics have relied on perception and acoustic analysis. The present study contributes to Colombian Spanish phonetics by investigating the articulation of sibilant fricatives. In particular, the shape of the palate and tongue during the production of sibilants is investigated in an attempt to quantify the shape and size of the oral cavity in the vicinity of the sibilant constriction. Real-time three-dimensional ultrasound, palate impressions, acoustic recordings, and electroglottography are brought to bear on these issues.

2aED11. Teaching acoustical interaction: An exploration of how teaching architectural acoustics to students spawns project-based learning. Daniel Butko, Haven Hardage, and Michelle Oliphant (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, Haven.B.Hardage-1@ou.edu)

The language and methods of architecture typically evaluated through small-scale models and drawings can be complemented by full-scale interactive constructs, augmenting learning through participatory, experiential, and sometimes experimental means. Congruent with Constantin Brancusi's proclamation, "architecture is inhabitable sculpture," opportunities to build full-scale constructs introduce students to a series of challenges predicated by structure, connections, safety, and a spirit of inquisition to learn from human interaction. To educate and entertain through sensory design, undergraduate students designed and built an interactive intervention allowing visual translation of acoustical impulses. The installation was developed and calibrated upon the lively acoustics and outward campus display of the college's gallery, employing excessive reverberation and resonance as a method of visually demonstrating sound waves. People physically inhabiting the space were the participants and critics by real-time reaction to personal interaction. The learning process complemented studio-based instruction through hands-on interaction with physical materials and elevated architectural education to a series of interactions with people. This paper documents and celebrates the Interactive Synchronicity project as a teaching tool outside common studio project representation while enticing classmates, faculty, and complete strangers to interact with inhabitable space.

2aED12. Palate shape and the central tongue groove. Coretta M. Talbert (Speech and Hearing Sci., Univ. of Southern MS, 211 Glen Court, Jackson, MS 39212, coretta.talbert@eagles.usm.edu) and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

It is well known that the center of the tongue can be grooved so that it is lower in the mouth than the lateral parts of the tongue, or it can bulge higher than the lateral parts of the tongue. It has never been shown whether or how this groove or bulge is related to the shape of the palate. In this study, we investigated the shape and size of the palate for several speakers using digitized 3D laser-scans of palate impressions and measurements on the impression plasters themselves. The groove or bulge in the center of the tongue was measured using real-time three-dimensional ultrasound. Pertinent findings will be presented concerning the relationship of the central groove/bulge shape and size to the shape and size of the palate.

2aED13. Signal processing for velocity and range measurement using a micromachined ultrasound transducer. Dominic Guri and Robert D. White (Mech. Eng., Tufts Univ., 200 College Ave., Anderson 204, Medford, MA 02155, dominic.guri@tufts.edu)

Signal processing techniques are under investigation for determination of range and velocity information from MEMS based ultrasound transducers. The ideal technique will be real-time, result in high resolution and accurate measurements, and operate successfully in noise. Doppler velocity measurements were previously demonstrated using a MEMS cMUT array (Shin *et al*., ASA Fall Meeting 2011, JASA 2013, Sens. Actuators A 2014). The MEMS array has 168 nickel-on-glass capacitive ultrasound transducers on a 1 cm die, and operates at 180 kHz in air. Post processing of the received ultrasound demonstrated the ability to sense velocity using continuous wave (CW) Doppler at a range of up to 1.5 m. The first attempt at real-time processing using a frequency modulated continuous wave (FM/CW) scheme was noise limited by the analog demodulation circuit. Further noise analysis is ongoing to determine whether this scheme may be viable. Other schemes under consideration include cross correlation chirp and single and multi-frequency burst waveforms. Preliminary results from a single frequency burst showed that cross-correlation-based signal processing may achieve acceptable range. The system is targeted at short range small robot navigation tasks. Determination of surface roughness from scattering of the reflected waves may also be possible.

2aED14. Investigation of a tongue-internal coordinate system for two-dimensional ultrasound. Rebecca Pedro, Elizabeth Mazzocco (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, rebpedro@indiana.edu), Tamás G. Csapó (Dept. of Telecommunications and Media Informatics, Budapest Univ. of Technol. and Economics, Budapest, Hungary), and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

In order to compare ultrasound recordings of tongue motion across utterances or across speakers, it is necessary to register the ultrasound images with respect to a common frame of reference. Methods for doing this typically rely either (1) on fixing the position of the ultrasound transducer relative to the skull by means of a helmet or a similar device, or (2) re-aligning the images by various means, such as optical tracking of head and transducer motion. These methods require sophisticated laboratory setups, and are less conducive to fieldwork or other studies in which such methods are impractical. In this study, we investigated the possibility of defining a rough coordinate system for image registration based on anatomical properties of the tongue itself. This coordinate system is anchored to the lower-jaw rather than the skull, but may potentially be transformed into an approximately skull-relative coordinate system by integrating video recordings of jaw motion.

2aED15. The effect of finite impedance ground reflections on horizontal full-scale rocket motor firings. Samuel Hord, Tracianne B. Neilsen, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., 737 N 600 E #103, Provo, UT 84606, samuel.hord@gmail.com)

Ground reflections have a significant impact on the propagation of sound from a horizontal rocket firing. The impedance of the ground relies strongly on effective flow resistivity of the surface and determines the frequencies at which interference nulls occur. For a given location, a softer ground, with lower effective flow resistivity, shifts the location of interference nulls to lower frequencies than expected for a harder ground. The difference in the spectral shapes from two horizontal firings of GEM-60 rocket motors, over snowy ground, clearly shows this effect and has been modeled. Because of the extended nature of high energy launch vehicles, the exhaust plume is modeled as a partially correlated line source, with distribution parameters chosen to match the recorded data sets as best as possible. Different flow resistivity values yield reasonable comparisons to the results of horizontal GEM-60 test firings.

2aED16. Palate-related constraints on sibilant production in three dimensions. Sarah Janssen and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, sejansse14@gmail.com)

Most studies of speech articulation are limited to a single plane, typically the midsagittal plane, although coronal planes are also used. Single-plane data have been undeniably useful in improving our understanding of speech production, but for many acoustic and aerodynamic processes, a knowledge of 3D vocal tract shapes is essential. In this study, we used palate impressions to investigate variations in the 3D structure of the palates of several individuals, and we used real-time 3D ultrasound to image the tongue surface during sibilant production by the same individuals. Our analysis focused on the degree to which tongue shapes during sibilant productions are substantially similar or different between individuals with different palate shapes and sizes.

2aED17. The evaluation of impulse response testing in low signal-to-noise ratio environments. Hannah D. Knorr (Audio Arts and Acoust., Columbia College Chicago, 134 San Carlos Rd, Address, Minooka, IL 60447, hknorr13@gmail.com), Jay Bleifnick (Audio Arts and Acoust., Columbia College Chicago, Schiller Park, IL), Andrew M. Hulva, and Dominique J. Chéenne (Audio Arts and Acoust., Columbia College Chicago, Chicago, IL)

Impulse testing is used by industry professionals to test many parameters of room acoustics, including the energy decay, frequency response, time response, etc. Current testing software makes this process as streamlined as possible, but generally must be utilized in quiet environments to yield high signal-to-ratios and more precise results. However, many real world situations cannot conform to the necessary standards needed for reliable data. This study tests various methods of impulse responses in background noise environments in an attempt to find the most reliable procedure for spaces with a high ambient noise levels. Additionally, extreme situations will be evaluated and a method will be derived to correct for the systematic error attributed to high background noise levels.

2aED18. Comparison of palate impressions and palate casts from three-dimensional laser-scanned digital models. Michelle Tebout and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, mtebout@imail.iu.edu)

Palate impressions and their casts in plaster are negatives of each other. While plaster casts are the standard for palate measurements and data preservation, making such casts can be time-consuming and messy. We hypothesized that measurements from 3D laser-scanned palate impressions are negligibly different from equivalent measurements from 3D laser-scanned palate casts. If true, this would allow the step of setting impressions in plaster to be skipped in future research. This poster presents the results of our study.

2aED19. The analysis of sound wave scattering using a firefighter's Personal Alert Safety System signal propagating through a localized region of fire. Andrew L. Broda (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402), Chase J. Rudisill (Phys. Dept., U.S. Naval Acad., Harwood, MD), Nathan D. Smith (Phys. Dept., U.S. Naval Acad., Davidsonville, MD), Matthew K. Schrader, and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD, korman@usna.edu)

Firefighting is quite clearly a dangerous and risk-filled job. To combat these dangers and risks, firefighters wear a (National Fire Protection Agency, NFPA 2007 edition of the 1982 standard) Personal Alert Safety System (PASS) that will sound a loud alarm if it detects (for example) the lack of movement of a firefighter. However, firefighters have experienced difficulty locating the source of these alarm chirps (95 dBA around 3 kHz) in a burning building. The project goal is to determine the effect of pockets of varying temperatures of air in a burning building on the sound waves produced by a PASS device. Sound scattering experiments performed with a vertical heated air circular jet plume (anechoic chamber) and with a wood fire plume from burning cylindrical containers (Anne Arundel Fire Department's Training Facility) suggest that from Snell's Law, sound rays refract around such pockets of warmer air surrounded by less warmer ambient air due to changes in the sound speed with temperature through the medium. Real-time and spectral measurements of 2.7 kHz CW sound scattering (using a microphone) exhibit some attenuation and considerable amplitude and frequency modulation. This research may suggest future experiments and effective modifications of the current PASS system.

2aED20. New phased array models for fast nearfield pressure simulations. Kenneth Stewart and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI, stewa584@msu.edu)

FOCUS, the "Fast Object-oriented C++ Ultrasound Simulator," is free MATLAB-based software that rapidly and accurately models therapeutic and diagnostic ultrasound with the fast nearfield method, time-space decomposition, and the angular-spectrum approach. FOCUS presently supports arrays of circular, rectangular, and spherically focused transducers arranged in flat planar, spherically focused, and cylindrically focused geometries. Excellent results are obtained with all of these array geometries in FOCUS for simulations of continuous-wave and transient excitations, and new array geometries are needed for B-mode simulations that are presently under development. These new array geometries also require new data structures that describe the electrical connectivity of the arrays. Efforts to develop these new features in FOCUS are underway, and results obtained with these new array geometries will be presented. Other new features for FOCUS will also be demonstrated. [Supported in part by NIH Grant R01 EB012079.]

2aED21. Nonlinear scattering of crossed focused ultrasonic beams in the presence of turbulence generated behind a model deep vein thrombosis using an orifice plate set in a thin tube. Daniel Fisher and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

An orifice plate (modeling a "blockage" in a deep vein thrombosis DVT) creates turbulent flow in a downstream region of a submerged polyethylene tube (1.6 mm thick, diameter 4 cm and overall length 40 cm). In the absence of the orifice plate, the water flow is laminar. The orifice plate is mechanically secured between two 20 cm tube sections connected by a union. The union allows a plate with an orifice to be slid into the union providing a concentric orifice plate that can obstruct the flow causing vorticity and turbulent flow downstream. A set of orifice plates (3 mm thick) are used (one at a time) to conveniently obstruct the tube flow with a different radius compared to the inner wall tube radius. The nonlinear scattering at the sum frequency (f_+ = 3.8 MHz), from mutually perpendicular spherical focused beams (f_1 = 1.8 MHz and f_2 = 2.0 MHz) is used to correlate the Doppler shift, spectral, and intensity as a function of the orifice plate size in an effort to correlate the blockage with the amount of nonlinear scattering. In the absence of turbulence in the overlap region, there is virtually no scattering. Therefore, a slight blockage is detectable.

2aED22. Analysis of acoustic data acquisition instrumentation for underwater blast dredging. Brenton Wallin, Alex Stott, James Hill, Timothy Nohara, Ryan Fullan, Jon Morasutti, Brad Clark, Alexander Binder, and Michael Gardner (Ocean Eng., Univ. of Rhode Island, 30 Summit Ave., Narragansett, RI 02882, brentwallin@my.uri.edu)

A team of seniors from the University of Rhode Island were tasked with analyzing the acoustic data and evaluating the data acquisition systems used in Pacific Northwest National Laboratories' (PNNL) study of blast dredging in the Columbia River. Throughout the semester, the students learned about the unique acoustic signatures of confined underwater blasts and the necessary specifications of systems used to record them. PNNL used two data acquisition systems. One was a tourmaline underwater blast sensor system created by PCB Piezotronics. The second was a hydrophone system using a Teledyne TC 4040 hydrophone, a Dytran inline charge amplifier, and a signal conditioner built for the blast sensor system. The students concluded that the data from the blast sensor system was reliable because the system was built by the company for this specific application and there were calibration sheets showing the system worked properly. The hydrophone data was deemed unreliable because components were orientated in an unusual manner that lead to improper data acquisition. A class of URI graduate students built a new hydrophone system that accurately recorded underwater dredge blasts performed in New York Harbor. This system is a fraction of the price of the blast sensor system.

2aED23. Effects of sustainable and traditional building systems on indoor environmental quality and occupant perceptions. Joshua J. Roberts and Lauren M. Ronsse (Audio Arts and Acoust., Columbia College Chicago, 4363 N. Kenmore Ave., Apt. #205, Chicago, IL 60613, joshua.roberts@loop.colum.edu)

This study examines the effects of both sustainable and traditional building systems on the indoor environmental quality (IEQ) and occupant perceptions in an open-plan office floor of a high-rise building located in Chicago, IL. The office evaluated has sustainable daylighting features as well as a more traditional variable air volume mechanical system. Different measurement locations and techniques are investigated to quantify the indoor environmental conditions (i.e., acoustics, lighting, and thermal conditions) experienced by the building occupants. The occupant perceptions of the indoor environmental conditions are assessed via survey questionnaires administered to the building occupants. The relationships between the IEQ measured in the office and the occupant perceptions are assessed.

TUESDAY MORNING, 28 OCTOBER 2014 — MARRIOTT 9/10, 8:00 A.M. TO 12:15 P.M.

Session 2aID

Archives and History and Engineering Acoustics: Historical Transducers

Steven L. Garrett, Chair
Grad. Prog. in Acoustics, Penn State, Applied Research Lab, P. O. Box 30, State College, PA 16804-0030

Chair's Introduction—8:00

Invited Papers

8:05

2aID1. 75th Anniversary of the Shure Unidyne microphone. Michael S. Pettersen (Applications Eng., Shure Inc., 5800 W. Touhy Ave., Niles, IL 60714, pettersen_michael@shure.com)

2014 marks the 75th anniversary of the Shure model 55 microphone. Introduced in 1939 and still being manufactured today, the Shure Unidyne was the first unidirectional microphone using a single dynamic element. The presentation provides an overview of the Unidyne's unique position in the history of 20th century broadcast, politics, and entertainment, plus the amazing story of Benjamin Bauer, a 24 year old immigrant from the Ukraine who invented the Unidyne and earned his first of over 100 patents for audio technology. Rare Unidyne artifacts from the Shure Archive will be on display after the presentation, including prototypes fabricated by Ben Bauer.

8:25

2aID2. Ribbon microphones. Wesley L. Dooley (Eng., Audio Eng. Assoc., 1029 North Allen Ave, Pasadena, CA 91104, wes@ribbonmics.com)

The ribbon microphone was invented by Dr. Walter Schottky who described it in German Patent 434855C, issued December 21, 1924 to Siemens & Halske (S&H) in Berlin. An earlier "Electro-Dynamic Loudspeaker" Patent which Schottky had written with Dr. Erwin Gerlach described a compliant, lightweight, and ribbed aluminum membrane whose thinnest dimension was at right angles to a strong magnetic field. Passing an audio frequency current through this membrane causes it to move and create sound vibrations. The December Patent describes how this design functions either as a loudspeaker or a microphone. A 1930 S&H patent for ribbon microphone improvements describes how they use internal resonant and ported chambers to extend frequency response past 4 kHz. RCA dramatically advanced ribbon microphone performance in 1931. They opened the ribbon to free air to create a consistent, air-damped, low-distortion, figure-eight with smooth 30–10,000 Hz response. RCA ribbon microphones became the performance leader for cinema, broadcast, live sound and recording. Their 20–30,000 Hz RCA 44B and BX was manufactured from 1936 to 1955. It is the oldest design still used every day at major studios. Ribbon microphones are increasingly used for contemporary recordings. Come hear why ribbon microphones, like phonograph records, are relevant to quality sound.

8:45

2aID3. Iconic microphonic moments in historic vocal recordings. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

Microphone selection—the strategic pairing of microphone make and model with each sound to be recorded—is one of the most important decisions a sound engineer must make. The technical specifications of the microphone identify which transducers are capable of functioning properly for any given recording task, but the ultimate decision is a creative one. The goal is for the performance capabilities of the microphone to not only address any practical recording session challenges, but also flatter the sound of the instrument, whether in pursuit of palpable realism or a fictionalized new timbre. The creative decision is informed, in part, by demonstrated success in prior recordings, the most important of which are described for that essential pop music instrument: the voice.

9:05

2aID4. The WE 640AA condenser microphone. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com)

In 1916 Edward Wente, working for Western Electric an AT&T subsidiary, invented a microphone that was the foundation of the modern condenser microphone. Wente's early condenser microphone designs continued to be developed until Western Electric produced the WE 361 in 1924 followed by the Model 394 condenser microphone in 1926. Western Electric used the WE 394 microphone as part of the "Master Reference System" to rate audio transmission quality of the telephone network. The WE 394 was too large for some measurement purposes so in 1932 Bell Labs engineers H. Harrison and P. Flanders designed a smaller version. The diaphragm had a diameter of 0.6 in. However, this design proved too difficult to manufacture and F. Romanow, also at Bell Labs, designed the 640A "1 in." microphone in 1932. Years later it was discovered that the 640A sensitivity varied by almost 6 dB from -650 C to 250 C. To reduce the thermal sensitivity, Bell Labs engineers M. Hawley and P. Olmstead carefully changed some of the 640A materials. The modified microphone was designated as the 640AA, which became the worldwide standard microphone for measuring sound pressure. This talk will describe some more details of the history of the 640AA microphone.

9:25

2aID5. Reciprocity calibration of condenser microphones. Leo L. Beranek (Retired, 10 Longwood Dr., Westwood, MA 02090, beranekleo@ieee.org)

The theory of reciprocity began with Lord Rayleigh and was first well stated by S. Ballantine (1929). The first detailed use of the reciprocity theory for the calibration of microphones was by R. K. Cook (1940). At the wartime Electro-Acoustic Laboratory, at Harvard University, the need arose to calibrate a large number of Western Electric 640-AA condenser microphones. A reciprocity apparatus was developed that connected the two microphones with an optimum shaped cavity that included a means for introducing hydrogen or helium to extend the frequency range. The apparatus was published by A. L. Dimattia and F. M. Wiener (1946). A number of things resulted. The Harvard group, in 1941, found that the international standard of sound pressure was off by 1.2 dB—that standard was maintained by the French Telephone Company and the Bell Telephone Laboratories and was based on measurements made with Thermophones. This difference was brought to the attention of those organizations and the reciprocity method of calibration was subsequently adopted by them resulting in the proper standard of sound pressure adopted around 1942. The one-inch condenser microphone has subsequently become the worldwide standard for precision measurement of sound field pressures.

9:45–10:00 Break

10:00

2aID6. Electret microphones. James E. West (ECE, Johns Hopkins Univ., 3400 N. Charles St., Barton Hall 105, Baltimore, MD 21218, jimwest@jhu.edu)

For nearly 40 years, condenser electret microphones have been the transducer of choice in most every area of acoustics including telephony, professional applications, hearing aids, and toys. More than 2 billion electret microphones are produced annually, primarily for the communications and entertainment markets. E. C. Wente invented the condenser microphone in 1917 at Bell Labs while searching for a replacement for the carbon microphone used in telephones; however, the necessary few hundred volt bias rendered the condenser microphone unusable in telephony, but its acoustical characteristics were welcomed in professional and measurement applications. Permanently charged polymers (electrets) provided the necessary few hundred-volt bias, thus simplifying the mechanical and electrical requirements for the condenser microphone and making it suitable for integration into the modern telephone. The introduction of inexpensive condenser microphones with matching frequency, phase, and impedance characteristics opened research opportunities for multiple microphone arrays. Array technology developed at Bell Labs will be presented in this talk.

10:20

2aID7. Meucci's telephone transducers. Angelo J. Campanella (Acculab, Campanella Assoc., 3201 Ridgewood Dr., Ohio, Hilliard, OH 43026, a.campanella@att.net)

Antonio Meucci (1809–1889) developed variable reluctance transducers from 1854 to 1876 and beyond after noticing that he could hear voice sounds from paddle electrodes while he participated in electrotherapy of a migraine patient around 1844 while in Havana, Cuba. He immigrated to Staten Island, NY, in 1850 and continued experimenting to develop a telephone. He found better success from electromagnetics using materials evolved from the telegraph developed by Morse, as well as a non-metal diaphragm with an iron tab, iron bars and a horseshoe shape. Artifacts from his residence on Staten Island are presently on display at the museum of his life on Staten Island from 1850 to his death. Those artifacts, thought until now to be only models, were found to be wired and still operative. Tests were performed in July, 2011. Their electrical resistance is that expected for wire wound variable reluctance transducers. Voice signals were produced without any externally supplied operating current. At least one transducer was found to be also operable as a receiver and was driven to produce voice sounds to the ear. Meucci's life and works will be discussed and these test results will be demonstrated including recordings from voice tests.

10:40

2aID8. The Fessenden Oscillator: The first sonar transducer. Thomas R. Howarth and Geoffrey R. Moss (U.S. Navy, 1176 Howell St, B1346 R404A, Newport, RI 02841, thomas.howarth@navy.mil)

When the RMS Titanic sunk in 1912, there was a call placed forth by ship owners for inventors to offer solutions for ship collision avoidance methods. Canadian born inventor Reginald A. Fessenden answered this call while working at the former Boston Submarine Signal Company with the invention and development of the first modern transducer used in a sonar. The Fessenden oscillator was an edge clamped circular metal with a radiating head facing the water on one side while the interior side had a copper tube attached that moved in and out of a fixed magnetic coil. The coil consisted of a direct-current (DC) winding to provide a magnetic field polarization and an alternating-current (AC) coil winding to induce the current into the copper tube and thus translate the magnetic field polarization to the radiating plate with vibrations that translated from the radiating head to the water medium. The prototype and early model versions operated at 540 Hz. Later developments included adaptation of this same transducer for use in underwater communications, obstacle avoidance with WW I retrofits onto British submarines for both transmitting and receiving applications including mine detection. This presentation will discuss design details including a modern numerical modelling effort.

11:00

2aID9. Historical review of underwater acoustic cylindrical transducer development in Russia for sonar arrays. Boris Aronov (ATMC/ECE, Univ. of Massachusetts Dartmouth, Needam, MA) and David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

Beginning with the introduction of piezoelectric ceramics in the 1950's, underwater acoustics transducer development for active sonar arrays proceeded in different directions in Russia (formerly USSR) than in the United States (US). The main sonar arrays in Russia were equipped with cylindrical transducers, whereas in the US, the implementation was most often made with extensional bar transducers of the classic Tonpilz design. The presentation focuses on the underlying objectives and humane factors that shaped the preference towards the widespread application of baffled cylindrical transducers for arrays in Russia, the history of their development, and contributions to theory of the transducers made by the pioneering developers.

11:20

2aID10. The phonodeik: Measuring sound pressure before electroacoustic transducers. Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., N-249 Millennium Sci. Complex, University Park, PA 16802, sct12@psu.edu)

The modern ability to visualize sound pressure waveforms using electroacoustic transducers began with the development of the vacuum tube amplifier, and has steadily improved as better electrical amplification devices have become available. Before electoral amplification was available; however, a significant body of acoustic pressure measurements had been made using the phonodeik, a device developed by Dayton C. Miller in the first decade of the twentieth century. The phonodeik employs acoustomechanical transduction to rotate a small mirror that reflects an optical beam to visualize the pressure waveform. This presentation will review the device and some of the discoveries made with it.

11:40

2aID11. A transducer not to be ignored: The siren. Julian D. Maynard (Phys., Penn State Univ., 104 Davey Lab, Box 231, University Park, PA 16802, maynard@phys.psu.edu)

An historic transducer to which one should pay attention is the siren. While its early application was as a source for a musical instrument, the siren soon became the transducer of choice for long-range audible warning because of its high intensity and recognizable tone. The components defining the siren include a solid stator and rotor, each with periodic apertures, and a compressed fluid (usually air but could be other fluids). With the rotor rotating in close proximity to the stator, and the resulting opening and closing of passageways through the apertures for the compressed fluid results in periodic sound waves in the surrounding fluid; usually a horn is used to enhance the radiation efficiency. The high potential energy of the compressed fluid permits high intensity sound. Some sirens which received scientific study include that of R. Clark Jones (1946), a 50 horsepower siren with an efficiency of about 70%, and that of C. H. Allen and I. Rudnick (1947), capable of ultrasonic frequencies and described as a "supersonic death ray" in the news media. Some design considerations, performance results, and applications for these sirens will be presented.

12:00–12:15 Panel Discussion

Session 2aMU

Musical Acoustics: Piano Acoustics

Nicholas Giordano, Chair
Physics, College of Sciences and Mathematics, Auburn University, Auburn, AL 36849

Invited Papers

9:00

2aMU1. The slippery path from piano key to string. Stephen Birkett (Systems Design Eng., Univ. of Waterloo, 250 University Ave., Waterloo, ON N2L 3G1, Canada, sbirkett@uwaterloo.ca)

Everything that contributes to the excitation of a piano string, from key input to hammer–string interaction, is both deterministic and consistently repeatable. Sequences of identical experimental trials give results that are indistinguishable. The simplicity of this behavior contrasts with the elusive goal of predicting input–output response and the extreme difficulty of accurate physical characterization. The nature and complexity of the mechanisms and material properties involved, as well as the sensitivity of their parameterization, place serious obstacles in the way of the usual investigative tools. This paper discusses and illustrates the limitations of modeling and simulation as applied to this problem, and the special considerations required for meaningful experimentation.

9:25

2aMU2. Coupling between transverse and longitudinal waves in piano strings. Nikki Etchenique, Samantha Collin, and Thomas R. Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, netchenique@rollins.edu)

It is known that longitudinal waves in piano strings noticeably contribute to the characteristic sound of the instrument. These waves can be induced by directly exciting the motion with a longitudinal component of the piano hammer, or by the stretching of the string associated with the transverse displacement. Longitudinal waves that are induced by the transverse motion of the string can occur at frequencies other than the longitudinal resonance frequencies, and the amplitude of the waves produced in this way are believed to vary quadratically with the amplitude of the transverse motion. We present the results of an experimental investigation that demonstrates the quadratic relationship between the magnitude of the longitudinal waves and the magnitude of the transverse displacement for steady-state, low-amplitude excitation. However, this relationship is only approximately correct under normal playing conditions.

9:50

2aMU3. Microphone array measurements, high-speed camera recordings, and geometrical finite-differences physical modeling of the grand piano. Rolf Bader, Florian Pfeifle, and Niko Plath (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

Microphone array measurements of a grand piano soundboard show similarities and differences between eigenmodes and forced oscillation patterns when playing notes on the instrument. During transients the driving point of the string shows enhanced energy radiation, still not as prominent as with the harpsichord. Lower frequencies are radiated stronger on the larger side of the soundboard wing shape, while higher frequencies are radiated stronger on the smaller side. A separate region at the larger part of the wing shape, caused by geometrical boundary conditions has a distinctly separate radiation behavior. High-speed camera recordings of the strings show energy transfer between strings of the same note. In physical models including hammer, strings, bridge, and soundboard the hammer movement is crucially necessary to produce a typical piano sound. Different bridge designs and bridge models are compared enhancing inharmonic sound components due to longitudinal-transversal coupling of the strings at the bridge.

10:15–10:35 Break

10:35

2aMU4. Adjusting the soundboard's modal parameters without mechanical change: A modal active control approach. Adrien Mamou-Mani (IRCAM, 1 Pl. Stravinsky, Paris 75004, France, adrien.mamou-mani@ircam.fr)

How do modes of soundboards affect the playability and the sound of string instruments? This talk will investigate this question experimentally, using modal active control. After identifying modal parameters of a structure, modal active control allows the adjustments of modal frequency and damping thanks to a feedback loop, without any mechanical changes. The potential of this approach for musical acoustics research will be presented for three different instruments: a simplified piano, a guitar, and a cello. The effects of modal active control of soundboards will be illustrated on attack, amplitude of sound partials, sound duration, playability, and "wolf tone" production.

11:00

2aMU5. Modeling the influence of the piano hammer shank flexibility on the sound. Juliette Chabassier (Magique 3D, Inria, 200 Ave. de la vieille tour, Talence 33400, France, juliette.chabassier@inria.fr)

A nonlinear model for a vibrating Timoshenko beam in non-forced unknown rotation is derived from the virtual work principle applied to a system of beam with mass at the end. The system represents a flexible piano hammer shank coupled to a hammer head. A novel energy-based numerical scheme is then provided and coupled to a global energy-preserving numerical solution for the whole piano (strings, soundboard, and sound propagation in the air). The obtained numerical simulations show that the pianistic touch clearly influences the spectrum of the piano sound of equally loud isolated notes. These differences do not come from a possible shock excitation on the structure, nor from a changing impact point, nor a "longitudinal rubbing motion" on the string, since neither of these features are modeled in our study.

Contributed Paper

11:25

2aMU6. Real-time tonal self-adaptive tuning for electronic instruments. Yijie Wang and Timothy Y. Hsu (School of Music, Georgia Inst. of Technol., 950 Marietta St. NW Apt 7303, Atlanta, GA 30318, yijiewang@gatech.edu)

A fixed tuning system cannot achieve just intonation on all intervals. A better approximation of just intonation is possible if the frequencies of notes are allowed to vary. Adaptive tuning is a class of methods that adjusts the frequencies of notes dynamically in order to maximize musical consonance. However, finding the optimal frequencies of notes directly based on some definition of consonance has shown to be difficult and computationally expensive. Instead, this paper proposes that the current key of the music is both a good summary of past notes and a good prediction of future notes, which can facilitate adaptive tuning. A method is proposed that uses a hidden Markov model to detect the current key of the music and compute optimal frequencies of notes based on the current key. In addition, a specialized online machine learning method that enforces symmetry among diatonic keys is presented, which can potentially adapt the model for different genres of music. The algorithm can operate in real time, is responsive to the notes played, and is applicable to various electronic instruments, such as MIDI pianos. This paper also presents comparisons between this proposed tuning system and conventional tuning systems.

TUESDAY MORNING, 28 OCTOBER 2014 MARRIOTT 3/4, 9:25 A.M. TO 11:35 A.M.

Session 2aNSa

Noise and Psychological and Physiological Acoustics: New Frontiers in Hearing Protection I

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

Elliott H. Berger, Cochair
Occupational Health & Environmental Safety Division, 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650

Chair's Introduction—9:25

Invited Papers

9:30

2aNSa1. How long are inexperienced-subjects "naive" for ANSI S12.6? Hilary Gallagher, Richard L. McKinley (Battlespace Acoust. Branch, Air Force Res. Lab., 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, richard.mckinley.1@us.af.mil), and Melissa A. Theis (ORISE, Air Force Res. Lab., Wright-Patterson AFB, OH)

ANSI S12.6-2008 describes the methods for measuring the real-ear attenuation of hearing protectors. Method A, trained-subject fit, was intended to describe the capabilities of the devices fitted by thoroughly trained users while Method B, inexperienced-subject fit, was intended to approximate the protection that can be attained by groups of informed users in workplace hearing conservation programs. Inexperienced subjects are no longer considered "naïve" according to ANSI S12.6 after 12 or more sessions measuring the attenuation of earplugs or semi-insert devices. However, an inexperienced subject that has received high quality video instructions may no longer be considered "naïve" or "inexperienced" even after just one session. AFRL conducted an ANSI S12.6-2008 Method B study to determine what effect, if any, high quality instructions had on the performance of naïve or inexperienced subjects and the number of trials where the subject could still be considered naïve or inexperienced. This experiment used ten subjects who completed three ANSI S12.6

measurements using the A-B-A training order and another 10 subjects completed the study using the B-A-B training order (A = high quality video instructions, B = short "earplug pillow-pack" written instructions). The attenuation results will be discussed and the implications for ANSI S12.6.

9:50

2aNSa2. Evaluation of variability in real-ear attenuation testing using a unique database—35 years of data from a single laboratory. Elliott H. Berger and Ronald W. Kieper (Personal Safety Div., 3M, 7911 Zionsville Rd., Indianapolis, IN 46268, elliott.berger@mmm.com)

The gold standard in measuring hearing protector attenuation since the late 1950s has been real-ear attenuation at threshold (REAT). Though well understood and standardized both in the U. S. (ANSI S3.19-1974 and ANSI S12.6-2008) and internationally (ISO 4869-1:1990), and known to provide valid and reliable estimates of protection for the test panel being evaluated, an area that is not clearly defined is the variability of the test measurements within a given laboratory. The test standards do provide estimates of uncertainty, both within and between laboratories, based on limited test data and interlaboratory studies, but thus far no published within-laboratory data over numerous tests and years have been available to provide empirical support for variability statements. This paper provides information from a one-of-a-kind database from a single laboratory that has conducted nearly 2500 studies over a period of 35 years in a single facility, managed by the same director (the lead author). Repeat test data on a controlled set of samples of a foam earplug, a premolded earplug, and two different earmuffs, with one of the data sets comprising 25 repeat tests over that 35-year period, will be used to demonstrate the inherent variability of this type of human-subject testing.

10:10

2aNSa3. Sound field uncertainty budget for real-ear attenuation at threshold measurement per ANSI S12.6 standards. Jeremie Voix and Céline Lapotre (École de technologie supérieure, Université du Québec, 1100 Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, jeremie.voix@etsmtl.ca)

In many national and international standards, the attenuation of Hearing Protection Devices is rated according to a psychophysical method called Real-Ear Attenuation at Threshold (REAT), which averages on a group of test-subjects the difference between the open and occluded auditory thresholds. In ANSI S12.6 standard, these REAT tests are conducted in a diffuse sound field in which sound uniformity and directionality are assessed by two objective microphone measurements. While the ANSI S12.6 standard defines these two criteria, it does not link the microphone measurements to the actual variation of sound pressure level at the eardrum that may originate from natural head movements during testing. This presentation examines this issue with detailed measurements conducted in an ANSI S12.6-compliant audiometric booth using an Artificial Test Fixture (ATF). The sound pressure level variations were recorded for movements of the ATF along the three main spatial axes and two rotation planes. From these measured variations and different head movements hypothetical scenarios, various sound field uncertainty budgets were computed. These findings will be discussed in order to eventually include them for uncertainty budget in a revised version of the ANSI S12.6 standard.

10:30

2aNSa4. Estimating effective noise dose when using hearing protection: Differences between ANSI S12.68 calculations and the auditory response measured with temporary threshold shifts. Hilary L. Gallagher, Richard L. McKinley (Battlespace Acoust., Air Force Res. Lab., AFRL/711HPW/RHCB, 2610 Seventh St, Wright-Patterson AFB, OH 45433-7901, hilary.gallagher.1@us.af.mil), Elizabeth A. McKenna (Ball Aerosp. and Technologies, Air Force Res. Lab., Wright-Patterson AFB, OH), and Mellisa A. Theis (ORISE, Air Force Res. Lab., Wright-Patterson AFB, OH)

ANSI S12.6 describes the methods for measuring the real-ear attenuation at threshold of hearing protectors. ANSI S12.68 describes the methods of estimating the effective A-weighted sound pressure levels when hearing protectors are worn. In theory, the auditory response, as measured by temporary threshold shifts (TTS), to an unoccluded ear noise exposure and an equivalent occluded ear noise exposure should produce similar behavioral results. In a series of studies conducted at the Air Force Research Laboratory, human subjects were exposed to continuous noise with and without hearing protection. Ambient noise levels during the occluded ear exposures were determined using ANSI S12.6 and ANSI S12.68. These equivalent noise exposures as determined by the ANSI S12.68 "gold standard" octave-band method produced significantly different auditory responses as measured with TTS. The methods and results from this study will be presented.

Contributed Papers

10:50

2aNSa5. Fit-testing, training, and timing—How long does it take to fit-test hearing protectors? Taichi Murata (Environ. Health Sci., Univ. of Michigan, School of Public Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226, ygo7@cdc.gov), Christa L. Themann, David C. Byrne, and William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH)

Hearing protector fit-testing is a Best Practice for hearing loss prevention programs and is gaining acceptance among US employers. Fit-testing quantifies hearing protector attenuation achieved by individual workers and ensures that workers properly fit and receive adequate protection from their protectors. Employers may be reluctant to conduct fit-testing because of expenses associated with worker time away from the job, personnel to administer the testing, and acquisition of a fit-test system. During field and laboratory studies conducted by the National Institute for Occupational Safety and Health (NIOSH), timing data for the fit-test process with the NIOSH HPD Well-Fit^{TM} system were analyzed. For workers completely naïve to fit-testing, the tests were completed within 15–20 minutes. Unoccluded test times were less than 4 minutes and occluded tests required less than 3 minutes. A significant learning effect was seen for the psychoacoustic method of adjustment used by HPD Well-Fit, explaining the shorter test times as subjects progressed through the unoccluded and occluded conditions. Most of the workers required about 5 minutes of training time. Test times and attenuations were tester-dependent, indicating the need to provide training to staff administering fit-tests in the workplace.

11:05

2aNSa6. Intra-subject fit variability using field microphone-in-real-ear attenuation measurement for foam, pre-molded and custom molded earplugs. Jeremie Voix (École de technologie supérieure, Université du Québec, 1100 Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, jeremie.voix@etsmtl.ca), Cecile Le Cocq (École de technologie supérieure, Université du Québec, Montreal, QC, Canada), and Elliott H. Berger (E•A•RCAL Lab, 3M Personal Safety Div., Indianapolis, IN)

In recent years, the arrival of several field attenuation estimation systems (FAES) on the industrial marketplace have enabled better assessment of hearing protection in real-life noise environments. FAES measure the individual attenuation of a given hearing protection device (HPD) as fitted by the end-user, but FAES enable predictions based only on measurements taken over a few minutes and do not account for what may occur later in the field over months or years as the earplug may be fitted slightly differently over time. This paper will use the field microphone-in-real-ear (F-MIRE) measurement technique to study in the laboratory how consistently a subject can fit and refit an HPD. A new metric, the intra-subject fit variability, will be introduced and quantified for three different earplugs (roll-down foam, premolded and custom molded), as fitted by two types of test subjects (experienced and inexperienced). This paper will present the experimental process used and statistical calculations performed to quantify intra-subject fit variability. As well, data collected from two different laboratories will be contrasted and reviewed as to the impact of trained versus untrained test subjects.

11:20

2aNSa7. A new perceptive method to measure active insertion loss of active noise canceling headsets or hearing protectors by matching the timbre of two audio signals. Remi Poncot and Pierre Guiu (Parrot S.A. France, 15 rue de montreuil, Paris 75011, France, poncotremi@gmail.com)

Attenuation of passive hearing protectors is assessed either by the Real Ear Attenuation at Threshold subjective method or by objective Measurements In the Real Ear. For Active Noise Cancelling headsets both methods do not practically apply. Alternative subjective methods based on loudness balance and masked hearing threshold techniques were proposed. However, they led to unmatched results with objective measurements at low frequency, diverging in either direction. Additionally, they are relatively long as frequency points of interest are measured one after the other. This paper presents a novel subjective method based on timbre matching, which has the originality of involving other perceptive mechanisms than the previous ones did. The attenuation performance of ANC headsets is rated by the change in pressure level of eight harmonics when the active noise reduction functionality is switched on. All harmonics are played at once, and their levels are adjusted by the test subject until he perceives the same timbre both in passive and active modes. A test was carried out by a panel of people in diffuse noise field conditions to assess the performance of personal consumer headphones. Early results show that the method is as repeatable as MIRE and lead to close results.

TUESDAY MORNING, 28 OCTOBER 2014 — INDIANA E, 8:15 A.M. TO 11:20 A.M.

Session 2aNSb

Noise and Structural Acoustics and Vibration: Launch Vehicle Acoustics I

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

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

Chair's Introduction—8:15

Invited Papers

8:20

2aNSb1. Inclusion of source extent and coherence in a finite-impedance ground reflection model with atmospheric turbulence. Kent L. Gee and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

Acoustic data collected in static rocket tests are typically influenced by ground reflections. Furthermore, the partial coherence of the ground interaction due to atmospheric turbulence can play a significant role for larger propagation distances. Because the rocket plume is an extended radiator whose directionality is the result of significant source correlation, assessment of the impact of ground reflections in the data must include these effects. In this paper, a finite impedance-ground, single-source interference approach [G. A. Daigle, J. Acoust. Soc. Am. 65, 45–49 (1979)] that incorporates both amplitude and phase variations due to turbulence is extended to distributions of correlated monopoles. The theory for obtaining the mean-square pressure from multiple correlated sources in the presence of atmospheric turbulence is described. The effects of source correlation and extent, ground effective flow resistivity, and turbulence parameters are examined in terms differences in relative sound pressure level across a realistic parameter space. Finally, the model prediction is compared favorably against data from horizontal firings of large solid rocket motors. [Work supported by NASA MSFC and Blue Ridge Research and Consulting, LLC.]

8:40

2aNSb2. Estimation of acoustic loads on a launch vehicle fairing. Mir Md M. Morshed (Dept. of Mech. Eng., Jubail Univ. College, Jubail Industrial City, Jubail 10074, Saudi Arabia, morshedm@ucj.edu.sa), Colin H. Hansen, and Anthony C. Zander (School of Mech. Eng., The Univ. of Adelaide, Adelaide, SA, Australia)

During the launch of space vehicles, there is a large external excitation generated by acoustic and structural vibration. This is due to acoustic pressure fluctuations on the vehicle fairing caused by the engine exhaust gases. This external excitation drives the fairing structure and produces large acoustic pressure fluctuations inside the fairing cavity. The acoustic pressure fluctuations not only produce high noise levels inside the cavity but also cause damage such as structural fatigue, and damage to, or destruction of, the payload inside the fairing. This is an important problem because one trend of the aerospace industry is to use composite materials for the construction of launch vehicle fairings, resulted in large-scale weight reductions of launch vehicles, but increased the noise transmission inside the fairing. This work investigates the nature of the external acoustic pressure distribution on a representative small launch vehicle fairing during liftoff. The acoustic pressure acting on a representative small launch vehicle fairing was estimated from the complex acoustic field generated by the rocket exhaust during liftoff using a non-unique source allocation technique which considered acoustic sources along the rocket engine exhaust flow. Numerical and analytical results for the acoustic loads on the fairing agree well.

9:00

2aNSb3. Prediction of acoustic environments from horizontal rocket firings. Clothilde Giacomoni and Janice Houston (NASA/MSFC, NASA Marshall Space Flight Ctr., Bldg 4203, Cube 3128, Msfc, AL 35812, clothilde.b.giacomoni@nasa.gov)

In recent years, advances in research and engineering have led to more powerful launch vehicles which yield acoustic environments potentially destructive to the vehicle or surrounding structures. Therefore, it has become increasingly important to be able to predict the acoustic environments created by these vehicles in order to avoid structural and/or component failure. The current industry standard technique for predicting launch-induced acoustic environments was developed by Eldred in the early 1970s. Recent work has shown Eldred's technique to be inaccurate for current state-of-the-art launch vehicles. Due to the high cost of full-scale and even sub-scale rocket experiments, very little rocket noise data is available. Much of the work thought to be applicable to rocket noise has been done with heated jets. A model to predict the acoustic environment due to a launch vehicle in the far-field was created. This was done using five sets of horizontally fired rocket data, obtained between 2008 and 2012. Through scaling analysis, it is shown that liquid and solid rocket motors exhibit similar spectra at similar amplitudes. This model is accurate for these five data sets within 5 dB of the measured data.

9:20

2aNSb4. Acoustics research of propulsion systems. Ximing Gao (NASA Marshall Space Flight Ctr., Atlanta, Georgia) and Janice Houston (NASA Marshall Space Flight Ctr., 650 S. 43rd St., Boulder, Colorado 80305, janice.d.houston@nasa.gov)

The liftoff phase induces high acoustic loading over a broad frequency range for a launch vehicle. These external acoustic environments are used in the prediction of the internal vibration responses of the vehicle and components. Present liftoff vehicle acoustic environment prediction methods utilize stationary data from previously conducted hold-down tests to generate 1/3 octave band Sound Pressure Level (SPL) spectra. In an effort to update the accuracy and quality of liftoff acoustic loading predictions, non-stationary flight data from the Ares I-X were processed in PC-Signal in two flight phases: simulated hold-down and liftoff. In conjunction, the Prediction of Acoustic Vehicle Environments (PAVE) program was developed in MATLAB to allow for efficient predictions of sound pressure levels (SPLs) as a function of station number along the vehicle using semi-empirical methods. This consisted of generating the Dimensionless Spectrum Function (DSF) and Dimensionless Source Location (DSL) curves from the Ares I-X flight data. These are then used in the MATLAB program to generate the 1/3 octave band SPL spectra. Concluding results show major differences in SPLs between the hold-down test data and the processed Ares I-X flight data making the Ares I-X flight data more practical for future vehicle acoustic environment predictions.

9:40

2aNSb5. Acoustics associated with liquid rocket propulsion testing. Daniel C. Allgood (NASA SSC, Bldg. 3225, Stennis Space Ctr., MS 39529, Daniel.C.Allgood@nasa.gov)

Ground testing of liquid rocket engines is a necessary step towards building reliable launch vehicles. NASA Stennis Space Center has a long history of performing both developmental and certification testing of liquid propulsion systems. During these test programs, the propulsion test article, test stand infrastructure and the surrounding community can all be exposed to significant levels of acoustic energy for extended periods of time. In order to ensure the safety of both personnel and equipment, predictions of these acoustic environments are conducted on a routine basis. This presentation will provide an overview of some recent examples in which acoustic analysis has been performed. Validation of these predictions will be shown by comparing the predictions to acoustic data acquired during small- and full-scale engine hot-fire testing. Applications of semi-empirical and advanced computational techniques will be reviewed for both sea-level and altitude test facilities.

10:00–10:20 Break

10:20

2aNSb6. Post-flight acoustic analysis of Epsilon launch vehicle at lift-off. Seiji Tsutsumi (JAXA's Eng. Digital Innovation Ctr., JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara, Kanagawa 252-5210, Japan, tsutsumi.seiji@jaxa.jp), Kyoichi Ui (Space Transportation Mission Directorate, JAXA, Tsukuba, Japan), Tatsuya Ishii (Inst. of Aeronautical Technol., JAXA, Chofu, Japan), Shinichiro Tokudome (Inst. of Space and Aeronautical Sci., JAXA, Sagamihara, Japan), and Kei Wada (Tokyo Office, Sci. Service Inc., Chuuou-ku, Japan)

Acoustic level both inside and outside the fairing is measured at the first Epsilon Launch Vehicle (Epsilon-1). The obtained data shows time-varying fluctuation due to the ascent of the vehicle. Equivalent stationary duration for such non-stationary flight data is determined based on the procedure described in NASA HDBK-7005. The launch pad used by the former M-V launcher is modified for the Epsilon based on the Computational Fluid Dynamics (CFD) and 1/42-scale model tests. Although the launch pad is compact and any water injection system is not installed, 10 dB reduction in overall sound pressure level (OASPL) is achieved due to the modification for the Epsilon, comparing with the M-V. Acoustic level inside the fairing satisfies the design requirement. Acoustic design of the launch pad developed here is revealed to be effective. Prediction of the acoustics level based on the Computational Fluid Dynamics (CFD) and subscale testing is also investigated by comparing with the flight measurement.

10:40

2aNSb7. Jet noise-based diagnosis of combustion instability in solid rocket motors. Hunki Lee, Taeyoung Park, Won-Suk Ohm (Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, ohm@yonsei.ac.kr), and Dohyung Lee (Agency for Defense Development, Daejeon, South Korea)

Diagnosis of combustion instability in a solid rocket motor usually involves *in-situ* measurements of pressure in the combustor, a harsh environment that poses challenges in instrumentation and measurement. This paper explores the possibility of remote diagnosis of combustion instability based on far-field measurements of rocket jet noise. Because of the large pressure oscillations associated with combustion instability, the wave process in the combustor has many characteristic features of nonlinear acoustics such as shocks and limit cycles. Thus the remote detection and characterization of instability can be performed by listening for the tell-tale signs of the combustor nonlinear acoustics, buried in the jet noise. Of particular interest is the choice of nonlinear acoustic measure (e.g., among skewness, bispectra, and Howell-Morfey Q/S) that best brings out the acoustic signature of instability from the jet noise data. Efficacy of each measure is judged against the static test data of two tactical motors (one stable, the other unstable).

11:00

2aNSb8. Some recent experimental results concerning turbulent coanda wall jets. Caroline P. Lubert (Mathematics & Statistics, James Madison Univ., 301 Dixie Ave., Harrisonburg, VA 22801, lubertcp@jmu.edu)

The Coanda effect is the tendency of a stream of fluid to stay attached to a convex surface, rather than follow a straight line in its original direction. As a result, in such jets mixing takes place between the jet and the ambient air as soon as the jet issues from its exit nozzle, causing air to be entrained. This air-jet mixture adheres to the nearby surface. Whilst devices employing the Coanda effect usually offer substantial flow deflection, and enhanced turbulence levels and entrainment compared with conventional jet flows, these prospective advantages are generally accompanied by significant disadvantages including a considerable increase in associated noise levels and jet breakaway. Generally, the reasons for these issues are not well understood and thus the full potential offered by the Coanda effect is yet to be realized. The development of a model for predicting the noise emitted by three-dimensional flows over Coanda surfaces would suggest ways in which the noise could be reduced or attenuated. In this paper, the results of recent experiments on a 3-D turbulent Coanda wall jet are presented. They include the relationship of SPL, shock cell distribution and breakaway to various flow parameters, and predictions of the jet boundary.

Session 2aPA

Physical Acoustics: Outdoor Sound Propagation

Kai Ming Li, Cochair
Mechanical Engineering, Purdue University, 140 South Martin Jischke, West Lafayette, IN 47907-2031

Shahram Taherzadeh, Cochair
Engineering & Innovation, The Open University, Walton Hall, Milton Keynes MK7 6AA, United Kingdom

Contributed Papers

8:30

2aPA1. On the inversion of sound fields above a locally reacting ground for direct impedance deduction. Kai Ming Li and Bao N. Tong (Mech. Eng., Purdue Univ., 177 South Russel St., West Lafayette, IN 47907-2099, mmkmli@purdue.edu)

A complex root-finding algorithm is typically used to deduce the acoustic impedance of a locally reacting ground by inverting the measured sound fields. However, there is an issue of uniquely determining the impedance from a measurement of an acoustic transfer function. The boundary loss factor F, which is a complex function, is the source of this ambiguity. It is associated with the spherical wave reflection coefficient Q for the reflected sound field. These two functions are dependent on a complex parameter known as the numerical distance w. The inversion of F leading to the multiple solutions of w can be identified as the root cause of the problem. To resolve this ambiguity, the zeroes and saddle points of F are determined for a given source/receiver geometry and a known acoustic impedance. They are used to establish the basins containing all plausible solutions. The topography of Q is further examined in the complex w-plane. A method for identifying the family of solutions and selecting the physically meaningful branch is proposed. Validation is provided by using numerical simulations as well as the experimentally data. The error and uncertainties in the deduced impedance are quantified.

8:45

2aPA2. An improved method for direct impedance deduction of a locally reacting ground. Bao N. Tong and Kai Ming Li (Mech. Eng., Purdue Univ., 177 South Russel St., West Lafayette, IN 47907-2099, bntong@purdue.edu)

An accurate deduction of the acoustic impedance of a locally reacting ground depends on a precise measurement of sound fields at short-ranges. However, measurement uncertainties exists in both the magnitude and the phase of the acoustic transfer function. By using the standard method, accurate determination of the acoustic impedance can be difficult when the measured phases become unreliable in many outdoor conditions. An improved technique, which relies only on the magnitude information, has been developed. A minimum of two measurements at two source/receiver configurations are needed to determine the acoustic impedance. Even in the absence of measurement uncertainties, a more careful analysis suggests that a third independent measurement is often needed to give an accurate solution. When experimental errors are inevitably introduced, a selection of optimal geometry becomes necessary to reduce the sensitivity of the deduced impedance to small variations in the data. A graphical method is provided which offers greater insight into the deduction of impedance and a downhill simplex algorithm has been developed to automate the procedure. Physical constraints are applied to limit the search region and to eliminate the rogue solutions. Several case studies using indoor and outdoor data are presented to validate the proposed technique.

9:00

2aPA3. Wavelet-like models for random media in wave propagation simulations. D. Keith Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Dev. Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), and Sergey N. Vecherin (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Dev. Ctr., Hanover, NH)

Simulations of wave propagation and scattering in random media are often performed by synthesizing the media from Fourier modes, in which the phases are randomized and the amplitudes tailored to provide a prescribed spectrum. Although this approach is computationally efficient, it cannot capture organization and intermittency in random media, which impacts higher-order statistical properties. As an alternative, we formulate a cascade model involving distributions of wavelet-like objects (quasi-wavelets or QWs). The QW model is constructed in a self-similar fashion, with the sizes, amplitudes, and numbers of offspring objects occurring at a constant ratios between generations. The objects are randomly distributed in space according to a Poisson process. The QW model is formulated in static (time-invariant), steady-state, and non-steady versions. Many diverse natural and man-made environments can be synthesized, including turbulence, porous media, rock distributions, urban buildings, and vegetation. The synthesized media can then be used in simulations of wave propagation and scattering.

9:15

2aPA4. Space-time correlation of acoustic signals in a turbulent atmosphere. Vladimir E. Ostashev, D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu), Sandra Collier (U.S. Army Res. Lab., Adelphi, MD), and Sylvain Cheinet (French-German Res. Inst. of Saint-Louis, Saint-Louis, France)

Scattering by atmospheric turbulence diminishes the correlation, in both space and time, of acoustic signals. This decorrelation subsequently impacts beamforming, averaging, and other techniques for enhancing signal-to-noise ratio. Space-time correlation can be measured directly with a phased microphone array. In this paper, a general theory for the space-time correlation function is presented. The atmospheric turbulence is modeled using the von Karman spatial spectra of temperature and wind velocity fluctuations and locally frozen turbulence (i.e., the Taylor's frozen turbulence hypothesis with convection velocity fluctuations). The theory developed is employed to calculate and analyze the spatial and temporal correlation of acoustic signals for typical regimes of an unstable atmospheric boundary layer, such as mostly cloudy or sunny conditions with light, moderate, or strong wind. The results obtained are compared with available experimental data.

9:30

2aPA5. Characterization of wind noise by the boundary layer meteorology. Gregory W. Lyons and Nathan E. Murray (National Ctr. for Physical Acoust., The Univ. of MS, 1 Coliseum Dr., University, MS 38677, gwlyons@go.olemiss.edu)

The fluctuations in pressure generated by turbulent motions of the atmospheric boundary layer are a principal noise source in outdoor acoustic measurements. The mechanics of wind noise involve not only stagnation pressure fluctuations at the sensor, but also shearing and self-interaction of turbulence throughout the flow, particularly at low frequencies. The contributions of these mechanisms can be described by the boundary-layer meteorology. An experiment was conducted at the National Wind Institute's 200-meter meteorological tower, located outside Lubbock, Texas in the Llano Estacado region. For two days, a 44-element 400-meter diameter array of unscreened NCPA-UMX infrasound sensors recorded wind noise continuously, while the tower and a Doppler SODAR measured vertical profiles of the boundary layer. Analysis of the fluctuating pressure with the meteorological data shows that the statistical structure of wind noise depends on both mean velocity distribution and buoyant stability. The root-mean-square pressure exhibits distinct scalings for stable and unstable stratification. Normalization of the pressure power spectral density depends on the outer scales. In stable conditions, the kurtosis of the wind noise increases with Reynolds number. Measures of noise intermittency are explored with respect to the meteorology.

9:45

2aPA6. Statistical moments for wideband acoustic signal propagation through a turbulent atmosphere. Jericho E. Cain (US Army Res. Lab., 1200 East West Hwy, Apt. 422, Silver Spring, MD 20910, jericho.cain@gmail.com), Sandra L. Collier (US Army Res. Lab., Adelphi, MD), Vladimir E. Ostashev, and David K. Wilson (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

Developing methods for managing noise propagation, sound localization, sound classification, and for designing novel acoustic remote sensing methods of the atmosphere requires a detailed understanding of the impact that atmospheric turbulence has on acoustic propagation. In particular, knowledge of the statistical moments of the sound field is needed. The first statistical moment corresponds to the coherent part of the sound field and it is need in beamforming applications. The second moment enables analysis of the mean intensity of a pulse in a turbulent atmosphere. Numerical solutions to a set of recently derived closed form equations for the first and second order statistical moments of a wideband acoustic signal propagating in a turbulent atmosphere with spatial fluctuations in the wind and temperature fields are presented for typical regimes of the atmospheric boundary layer.

10:00–10:15 Break

10:15

2aPA7. Analysis of wind noise reduction by semi-porous fabric domes. Sandra L. Collier (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783-1197, sandra.l.collier4.civ@mail.mil), Richard Raspet (National Ctr. for Physical Acoust., Univ. of MS, University, MS), John M. Noble, W. C. Kirkpatrick Alberts (U.S. Army Res. Lab., Adelphi, MD), and Jeremy Webster (National Ctr. for Physical Acoust., Univ. of MS, University, MS)

For low frequency acoustics, the wind noise contributions due to turbulence may be divided into turbulence–sensor, turbulence–turbulence, and turbulence–mean shear interactions. Here, we investigate the use of a semi-porous fabric dome for wind noise reduction in the infrasound region. Comparisons are made between experimental data and theoretical predictions from a wind noise model [Raspet, Webster, and Naderyan, J. Acoust. Soc. Am. **135**, 2381 (2014)] that accounts for contributions from the three turbulence interactions.

10:30

2aPA8. An investigation of wind-induced and acoustic-induced ground motions. Vahid Naderyan, Craig J. Hickey, and Richard Raspet (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, NCPA, 1 Coliseum Dr.,, University, MS 38677, vnaderya@go.olemiss.edu)

Wind noise at low frequency is a problem in seismic surveys, which reduces seismic image clarity. In order to find a solution for this problem, we investigated the driving pressure perturbations on the ground surface associated with wind-induced ground motions. The ground surface pressure and shear stress at the air–ground interface were used to predict the displacement amplitudes of the horizontal and vertical ground motions as a function of depth. The measurements were acquired at a site having a flat terrain and low seismic ambient noise under windy conditions. Multiple triaxial geophones were deployed at different depths to study the induced ground velocity as a function of depth. The measurements show that the wind excites horizontal components more than vertical component on the above ground geophone due to direct interaction with the geophone. For geophones buried flush with the ground surface and at various depths below the ground, the vertical components of the velocity are greater than the horizontal components. There is a very small decrease in velocity with depth. The results are compared to acoustic-ground coupling case. [This work is supported by USDA under award 58-6408-1-608.]

10:45

2aPA9. Using an electro-magnetic analog to study acoustic scattering in a forest. Michelle E. Swearingen (US Army ERDC, Construction Eng. Res. Lab., P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil) and Donald G. Albert (US Army ERDC, Hanover, NH)

Using scale models can be a convenient method for investigating multiple scattering in complex environments, such as a forest. However, the increased attenuation with increasing frequency limits the propagation distances available for such models. An electromagnetic analog is an alternative way to study multiple scattering from rigid objects, such as tree trunks. This analog does not suffer from the intrinsic attenuation and allows for investigation of a larger effective area. In this presentation, the results from a 1:50 scale electromagnetic analog are compared to full-scale data collected in a forest. Further tests investigate propagation along multiple paths through a random configuration of aluminum cylinders representing trees. Special considerations and anticipated range of applicability of this analog method are discussed.

11:00

2aPA10. Modeling of sound scattering by an obstacle located below a hardbacked rigid porous medium. Yiming Wang and Kai Ming Li (Mech. Eng., Purdue Univ., 177 South Russel St., West Lafayette, IN 47907-2031, mmkmli@purdue.edu)

The boundary integral equation (BIE) formulation takes advantage of the well-known Green's function for the sound fields above a plane interface. It can then lead to a simplified numerical solution known as the boundary element method (BEM) that enables an accurate computation of sound fields above the plane interface with the presence of obstacles of complex shapes. The current study is motivated by the need to explore the acoustical characteristics of a layer of sound absorption materials embedded with equally spaced rigid inserts. In principle, this problem may be solved by a standard finite element program but it is found more efficient to use the BIE approach by discretizing only the boundary surfaces of the obstacles within the medium. The formulation is facilitated by using accurate Green's functions for computing the sound fields above and within a layer of rigid porous medium. This paper reports a preliminary study to model the scattering of sound by an obstacle placed within the layered rigid porous medium. The two-dimensional Green's functions will be derived and used for the development of a BEM model for computing the sound field above and within the rigid porous medium due to the presence of an arbitrarily shaped obstacle.

11:15

2aPA11. Analysis of the Green's function for a duct and cavity using geometric image sources. Ambika Bhatta, Charles Thompson, and Kavitha Chandra (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, ambika_bhatta@student.uml.edu)

The presented work investigates the solution for pressure response of a point source in a two dimensional waveguide. The methodology is based on the one dimensional analytical and numerical solution of a finite channel response between two semi-infinite planes. The branch integrals representing the reflection coefficient is implemented to evaluate the pressure amplitude of the boundary effect. The approach addresses the validation of application of geometric image sources for finite boundaries. Consequently, the 3D extension of the problem for a closed cavity is also investigated.

TUESDAY MORNING, 28 OCTOBER 2014 MARRIOTT 1/2, 8:00 A.M. TO 10:00 A.M.

Session 2aSAa

Structural Acoustics and Vibration and Noise: Computational Methods in Structural Acoustics and Vibration

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

Matthew Kamrath, Cochair
Acoustics, Pennsylvania State University, 717 Shady Ridge Road, Hutchinson, MN 55350

Invited Papers

8:00

2aSAa1. A radical technology for modeling target scattering. David Burnett (Naval Surface Warfare Ctr., Code CD10, 110 Vernon Ave., Panama City, FL 32407, david.s.burnett@navy.mil)

NSWC PCD has developed a high-fidelity 3-D finite-element (FE) modeling system that computes acoustic color templates (target strength vs. frequency and aspect angle) of single or multiple realistic objects (e.g., target + clutter) in littoral environments. High-fidelity means that 3-D physics is used in all solids and fluids, including even thin shells, so that solutions include not only all propagating waves but also all evanescent waves, the latter critically affecting the former. Although novel modeling techniques have accelerated the code by several orders of magnitude, NSWC PCD is now implementing a radically different FE technology, e.g., one thin-shell element spanning 90° of a cylindrical shell. It preserves all the 3-D physics but promises to accelerate the code another two to three orders of magnitude. The talk will briefly review the existing system and then describe the new technology.

8:20

2aSAa2. Faster frequency sweep methods for structural vibration and acoustics analyses. Kuangcheng Wu (Ship Survivability, Newport News ShipBldg., 202 Schembri Dr., Yorktown, VA 23693, kc.wu@hii-nns.com) and Vincent Nguyen (Ship Survivability, Newport News ShipBldg., Newport News, VA)

The design of large, complex structures typically requires knowledge of the mode shape and forced response near major resonances to ensure deflection, vibration, and the resulting stress are kept below acceptable levels, and to guide design changes where necessary. Finite element analysis (FEA) is commonly used to predict Frequency Response Functions (FRF) of the structure. However, as the complexity and detail of the structure grows, the system matrices, and the computational resources needed to solve them, get large. Furthermore, the need to use small frequency steps to accurately capture the resonant response peaks can drive up the number of FRF calculations required. Thus, the FRF calculation can be computationally expensive for large structural systems. Several approaches have been proposed that can significantly accelerate the overall process by approximating the frequency dependent response. Approximation approaches based on Krylov Galerkin Projection (KGP) and Pade calculate the forced response at only a few frequencies, then use the response and its derivatives to reconstruct the FRF in-between the selected direct calculation points. This paper first validates the two approaches with analytic solutions for a simply supported plate, and then benchmarks several numerical examples to demonstrate the accuracy and efficiency of the new approximate methods.

8:40

2aSAa3. Waves in continua with extreme microstructures. Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu)

The effective properties of a material may be generally defined as those that describe the limiting case where the wavelength of propagation is infinite compared to the characteristic scale of the microstructure. Generally, the limit of vanishingly small microstructural scale in a heterogeneous elastic medium results in an effective homogeneous medium that is again elastic. We show that for materials with extreme microstructures, the limiting effective medium can be quite exotic, including polar materials, or multiphase continuum. These continuum models naturally give rise to unusual effective properties including negative or anisotropic mass. Though unusual, these properties have straightforward interpretations in terms of the laws of classical mechanics. Finally, we discuss wave propagation in these structures and find dispersion curves with multiple branches.

Contributed Papers

9:00

2aSAa4. A comparison of perfectly matched layers and infinite elements for exterior Helmholtz problems. Gregory Bunting (Computational Solid Mech. and Structural Dynam., Sandia National Labs., 709 Palomas Dr. NE, Albuquerque, NM 87108, bunting.gregory@gmail.com), Arun Prakash (Lyles School of Civil Eng., Purdue Univ., West Lafayette, IN), and Timothy Walsh (Computational Solid Mech. and Structural Dynam., Sandia National Labs., West Lafyette, IN)

Perfectly matched layers and infinite elements are commonly used for finite element simulations of acoustic waves on unbounded domains. Both involve a volumetric discretization around the periphery of an acoustic mesh, which itself surrounds a structure or domain of interest. Infinite elements have been a popular choice for these problems since the 1970s. Perfectly matched layers are a more recent technology that is gaining popularity due to ease of implementation and effectiveness as an absorbing boundary condition. In this study, we present massively parallel implementations of these two techniques, and compare their performance on a set of representative structural–acoustic problems on exterior domains. We examine the conditioning of the linear systems generated by the two techniques by examining the number of Krylov-iterations needed for convergence to a fixed solver tolerance. We also examine the effects of PML parameters, exterior boundary conditions, and quadrature rules on the accuracy of the solution. [Sandia National Laboratories is a multi-program laboratory managed and operated by Sandia Corporation, a wholly owned subsidiary of Lockheed martin Corporation, for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-AC04-94AL850000.]

9:15

2aSAa5. Improved model for coupled structural–acoustic modes of tires. Rui Cao, Nicholas Sakamoto, and J. S. Bolton (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 177 S. Russell St., West Lafayette, IN 47907-2099, cao101@purdue.edu)

Experimental measurements of tire tread band vibration have provided direct evidence that higher order structural-acoustic modes exist in tires, not just the well-know fundamental mode. These modes display both circumferential and radial pressure variations. The theory governing these modes has thus been investigated. A brief recapitulation of the previously-presented coupled tire-acoustical model based on a tensioned membrane approach will be given, and then an improved tire-acoustical model with a ring-like shape will be introduced. In the latter model, the effects of flexural and circumferential stiffness are considered, as is the role of curvature in coupling the various wave types. This improved model accounts for propagating in-plane vibration in addition to the essentially structure-borne flexural wave and the essentially airborne longitudinal wave accounted for in the previous model. The longitudinal structure-borne wave "cuts on" at the tire's circumferential ring frequency. Explicit solutions for the structural and acoustical modes will be given in the form of dispersion relations. The latter results will be compared with measured dispersion relations, and the features associated primarily with the higher order acoustic modes will be highlighted. Finally, the effect of tire rotational speed on the natural frequencies of these various modes types will also be discussed.

9:30

2aSAa6. Simulating sound absorption in porous material with the lattice Boltzmann method. Andrey R. da Silva (Ctr. for Mobility Eng., Federal Univ. of Santa Catarina, Rua Monsenhor Topp, 173, Florianópolis, Santa Catarina 88020-500, Brazil, andrey.rs@ufsc.br), Paulo Mareze, and Eric Brandão (Structure and Civil Eng., Federal Univ. of Santa Maria, Santa Maria, RS, Brazil)

The development of porous materials that are able to absorb sound in specific frequency bands has been an important challenge in the acoustic research. Thus, the development new numerical techniques that allow one to correctly capture the mechanisms of sound absorption can be seen as an important step to developing new materials. In this work, the lattice Boltzmann method is used to predict the sound absorption coefficient in porous material with straight porous structure. Six configurations of porous material were investigated, involving different thickness and porosity values. A very good agreement was found between the numerical results and those obtained by the analytical model provided in the literature. The results suggest that the lattice Boltzmann model can be a powerful alternative to simulating viscous sound absorption, particularly due to its reduced computational effort when compared to traditional numerical methods.

9:45

2aSAa7. Energy flow models for the out-of-plane vibration of horizontally curved beams. Hyun-Gwon Kil (Dept. of Mech. Eng., Univ. of Suwon, 17, Wauan-gil, Bongdam-eup, Hwaseong-si, Gyeonggi-do 445-743, South Korea, hgkil@suwon.ac.kr), Seonghoon Seo (Noise & Vib. CAE Team, Hyundai Motor Co., Hwaseong-si, Gyeonggi-do, South Korea), Suk-Yoon Hong (Dept. of Naval Architecture and Ocean Eng., Seoul National Univ., Seoul, South Korea), and Chan Lee (Dept. of Mech. Eng., Univ. of Suwon, Hwaseong-si, Gyeonggi-do, South Korea)

The purpose of this work is to develop energy flow models to predict the out-of-plane vibration of horizontally curved beams in the mid- and high-frequency range. The dispersion relations of waves are approximately separated into relations to the propagation of flexural waves and torsional waves generating the out-of-plane vibration of the horizontally curved beams with Kirchhoff-Love hypotheses. The energy flow models are based on the energy governing equations for the flexural waves and the torsional waves propagating in the curved beams. Those equations are driven to predict the time- and locally space-averaged energy density and intensity in the curved beams. Total values for the energy density and the intensity as well as contributions of each type of waves on those values are predicted. A verification of the energy flow models for the out-of-plane vibration of the horizontally curved beams is performed by comparing the energy flow solutions for the energy density and the intensity with analytical solutions evaluated using the wave propagation approach. The comparison shows that the energy flow models can be effectively used to predict the out-of-plane vibration of the horizontally curved beams in the mid- and high-frequency range.

Session 2aSAb

Structural Acoustics and Vibration and Noise: Vehicle Interior Noise

Sean F. Wu, Chair
Mechanical Engineering, Wayne State University, 5050 Anthony Wayne Drive, College of Engineering Building, Rm. 2133, Detroit, MI 48202

Chair's Introduction—10:30

Invited Papers

10:35

2aSAb1. Structural–acoustic optimization of a pressurized, ribbed aircraft panel. Micah R. Shepherd and Stephen A. Hambric (Appl. Res. Lab, Penn State Univ., PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu)

A method to reduce the noise radiated by a ribbed, aircraft panel excited by turbulent boundary layer flow is presented. To compute the structural-acoustic response, a modal approach based on finite element/boundary element analysis was coupled to a turbulent boundary flow forcing function. A static pressure load was also applied to the panel to simulate cabin pressurization during flight. The radiated sound power was then minimized by optimizing the horizontal and vertical rib location and rib cross section using an evolutionary search algorithm. Nearly 10 dB of reduction was achieved by pushing the ribs to the edge of the panel, thus lowering the modal amplitudes excited by the forcing function. A static constraint was then included in the procedure using a low-frequency dynamic calculation to approximate the static response. The constraint limited the amount of reduction that was achieved by the optimizer.

11:00

2aSAb2. Extending interior near-field acoustic holography to visualize three-dimensional objective parameters of sound quality. Huancai Lu (Mech. Eng., Zhejiang Univ. of Technol., 3649 Glenwood Ave., Windsor, ON N9E 2Y6, Canada, huancailu@zjut.edu.cn)

It is essential to understand that the ultimate goal of interior noise control is to improve the sound quality inside the vehicle, rather than to suppress the sound pressure level. Therefore, the vehicle interior sound source localization and identification should be based on the contributions of sound sources to the subjective and/or objective parameters of sound quality at targeted points, such as driver's ear positions. This talk introduces the visualization of three-dimensional objective parameters of sound quality based on interior near-field acoustic holography (NAH). The methodology of mapping three-dimensional sound pressure distribution, which is reconstructed based on interior NAH, to three-dimensional loudness is presented. The mathematical model of loudness developed by ANSI standard is discussed. The numerical interior sound field, which is generated by vibrating enclosure with known boundary conditions, is employed to validate the methodology. In addition, the accuracy of reconstruction of loudness distribution is examined with ANSI standard and digital head. It is shown that the results of sound source localization based on three-dimensional loudness distribution are different from the ones based on interior NAH.

Contributed Paper

11:25

2aSAb3. A comparative analysis of the Chicago Transit Authority's Red Line railcars. Chris S. Nottoli (Riverbank Acoust. Labs., 1145 Walter, Lemont, IL 60439, cnottoli18@gmail.com)

A noise study was conducted on Chicago Transit Authority's Red Line railcars to assess the differences in interior sound pressure level between the 5000 series railcars and its predecessor, the 2400 series. The study took into account potential variability associated with a rider's location in the railcars, above ground, and subway segments (between stations), and surveyed the opinion of everyday Red Line riders as pertaining to perceived noise. The test data demonstrated a 3–6 dB noise reduction in ongoing CTA renovations between new rapid transit cars and their predecessors. Location on the train influenced Leq(A) measurements as reflections from adjacent railcars induced higher noise levels. The new railcars also proved effective in noise reduction throughout the subway segments as the averaged Leq(A) deviated 1 dB from above ground rail stations. Additionally, this study included an online survey that revealed a possible disconnect between traditional methods of objective noise measurement and subjective noise ratings.

Session 2aSC

Speech Communication: Speech Production and Articulation (Poster Session)

Sam Tilsen, Chair
Cornell University, 203 Morrill Hall, Ithaca, NY 14853

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and authors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

2aSC1. Tongue motion characteristics during vowel production in older children and adults. Jennell Vick, Michelle Foye (Psychol. Sci., Case Western Reserve Univ., 11635 Euclid Ave., Cleveland, OH 44106, jennell@case.edu), Nolan Schreiber, Greg Lee (Elec. Eng. Comput. Sci., Case Western Reserve Univ., Cleveland, OH), and Rebecca Mental (Psychol. Sci., Case Western Reserve Univ., Cleveland, OH)

This study examined tongue movements in consonant-vowel-consonant sequences drawn from real words in phrases as produced by 36 older children (three male and three female talkers at each age from 10 to 15 years) and 36 adults. Movements of four points on the tongue were tracked at 400 Hz using the Wave Electromagnetic Speech Research System (NDI, Waterloo, ON, CA). The four points were tongue tip (TT; 1 cm from tip on midline), tongue body (TB; 3 cm from tip on midline), tongue right (TR; 2 cm from tip on right lateral edge), and tongue left (TR; 2 cm from tip on left lateral edge). The phrases produced included the vowels /i/, /I/, /ae/, and /u/ in words (i.e., "see," sit," cat," and "zoo"). Movement measures included 3D distance, peak and average speed, and duration of vowel opening and closing strokes. The horizontal curvature of the tongue was calculated at the trajectory speed minimum associated with the vowel production using a least-squares quadratic fit of the TR, TB, and TL positional coordinates. Symmetry of TR and TL vertical position was also calculated. Within-group comparisons were made between vowels and between-group comparisons were made between children and adults.

2aSC2. Experimental evaluation of the constant tongue volume hypothesis. Zisis Iason Skordilis, Vikram Ramanarayanan (Signal Anal. and Interpretation Lab., Dept. of Elec. Eng., Univ. of Southern California, 3710 McClintock Ave., RTH 320, Los Angeles, CA 90089, skordili@usc.edu), Louis Goldstein (Dept. of Linguist, Univ. of Southern California, Los Angeles, CA), and Shrikanth S. Narayanan (Signal Anal. and Interpretation Lab., Dept. of Elec. Eng., Univ. of Southern California, Los Angeles, CA)

The human tongue is considered to be a muscular hydrostat (Kier and Smith, 1985). As such, it is considered to be incompressible. This constant volume hypothesis has been incorporated in various mathematical models of the tongue, which attempt to provide insights into its dynamics (e.g., Levine *et al.*, 2005). However, to the best of our knowledge, this hypothesis has not been experimentally validated for the human tongue during actual speech production. In this work, we attempt an experimental evaluation of the constant tongue volume hypothesis. To this end, volumetric structural Magnetic Resonance Imaging (MRI) was used. A database consisting of 3D MRI images of subjects articulating continuants was considered. The subjects sustained contextualized vowels and fricatives (e.g., IY in "beet," F in "afa") for 8 seconds in order for the 3D geometry to be collected. To segment the tongue and estimate its volume, we explored watershed (Meyer and Beucher, 1990) and region growing (Adams and Bischof, 1994) techniques. Tongue volume was estimated for each lingual posture for each subject. Intra-subject tongue volume variation was examined to determine if there is sufficient statistical evidence for the validity of the constant volume hypothesis. [Work supported by NIH and a USC Viterbi Graduate School Ph.D. fellowship.]

2aSC3. A physical figure model of tongue muscles. Makoto J. Hirayama (Faculty of Information Sci. and Technol., Osaka Inst. of Technol., 1-79-1 Kitayama, Hirakata 573-0196, Japan, mako@is.oit.ac.jp)

To help understanding tongue shape and motions, a physical figure model of tongue muscles using viscoelastic material of urethane rubber gel were made by improving previous models. Compare to previous shape tongue models that had been made and presented, the new model is constructed from tongue body (consisting of Transversus linguae, Verticalis linguae, Longitudinalis linguae superior, and Longitudinalis linguae inferior), and individual extrinsic tongue muscles (consisting of Genioglossus anterior, Genio glossus posterior, Hyoglossus, Styloglossus, and Palatoglossus) parts. Therefore, each muscle's shape, starting and ending points, and relation to other muscles and organs inside mouth are more understandable than previous ones. As the model is made from viscoelastic material similar to human skin, reshaping and moving tongue are possible by pulling or pushing some parts of the tongue muscle by hand, that is, tongue shape and motion simulations by hand can be done. The proposed model is useful for speech science education or a future speaking robot using realistic speech mechanism.

2aSC4. Tongue width at rest versus tongue width during speech: A comparison of native and non-native speakers. Sunao Kanada and Ian Wilson (CLR Phonet. Lab, Univ. of Aizu, Tsuruga, Ikki machi, Aizuwkamatsu, Fukushima 965-8580, Japan, m5181137@u-aizu.ac.jp)

Most pronunciation researchers do not focus on the coronal view. However, it is also important to observe because the tongue is hydrostatic. We believe that some pronunciation differences between native speakers and second-language (L2) speakers could be due to differences in the coronal plane. Understanding these differences could be a key to L2 learning and modeling. It may be beneficial for pedagogical purposes and the results of this research may contribute to the improvement of pronunciation of L2 English speakers. An interesting way to look at native and L2 articulation differences is through the pre-speech posture and inter-speech posture (ISP—rest position between sentences). In this research, we compare native speakers to L2 speakers. We measure how different those postures are from the median position of the tongue during speech. We focus on movement of a side tongue marker in the coronal plane, and we normalize for speaker size. We found that the mean distance from pre-speech posture to speech posture is shorter for native English speakers (0.95 mm) than for non-native English speakers (1.62 mm). So, native speakers are more efficient in their pre-speech posture. Results will also be shown for distances from ISP to speech posture.

2aSC5. Intraglottal velocity and pressure measurements in a hemilarynx model. Liran Oren, Sid Khosla (Otolaryngol., Univ. of Cincinnati, PO Box 670528, Cincinnati, OH 45267, orenl@ucmail.uc.edu), and Ephraim Gutmark (Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH)

Determining the mechanisms of self-sustained oscillation of the vocal folds requires characterization of intraglottal aerodynamics. Since most of the intraglottal aerodynamics forces cannot be measured in a tissue model of the larynx, most of the current understanding of vocal fold vibration mechanism is derived from mechanical, analytical, and computational models. In the current study, intraglottal pressure measurements are taken in a hemilarynx model and are compared with pressure values that are computed form simultaneous velocity measurements. The results show that significant negative pressure is formed near the superior aspect of the folds during closing, which is in agreement with previous measurements in a hemilarynx model. Intraglottal velocity measurements show that the flow near the superior aspect separates from the glottal wall during closing and may develop into a vortex, which further augments the magnitude of the negative pressure. The intraglottal pressure distributions are computed by solving the pressure Poisson equation using the velocity field measurements and show good agreement with the pressure measurements. The match between the pressure computations and the pressure measurements validates the technique, which was also used in previous study to estimate the intraglottal pressure distribution in a full larynx model.

2aSC6. Ultrasound study of diaphragm motion during tidal breathing and speaking. Steven M. Lulich, Marguerite Bonadies (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu), Meredith D. Lulich (Southern Indiana Physicians, Indiana Univ. Health, Bloomington, IN), and Robert H. Withnell (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

Studies of speech breathing by Ladefoged and colleagues (in the 1950s and 1960s), and by Hixon and colleagues (in the 1970s, 1980s, and 1990s) have substantially contributed to our understanding of respiratory mechanics during speech. Even so, speech breathing is not well understood when contrasted with phonation, articulation, and acoustics. In particular, diaphragm involvement in speech breathing has previously been inferred from inductive plethysmography and EMG, but it has never been directly investigated. In this case study, we investigated diaphragm motion in a healthy adult male during tidal breathing and conversational speech using real-time 3D ultrasound. Calibrated inductive plethysmographic data were recorded simultaneously for comparison with previous studies and in order to relate lung volumes directly to diaphragm motion.

2aSC7. A gestural account of Mandarin tone sandhi. Hao Yi and Sam Tilsen (Dept. of Linguist, Cornell Univ., 315-7 Summerhill Ln., Ithaca, NY 14850, hy433@cornell.edu)

Recently tones have been analyzed as articulatory gestures, which may be coordinated with segmental gestures. Our data from electromagnetic articulometry (EMA) show that purported neutralized phonological contrast can nonetheless exhibit coordinative difference. We develop a model based on gestural coupling to account for observed patterns. Mandarin Third Tone Sandhi (e.g., Tone3 → T3S /_ Tone3) is perceptually neutralizing in that the sandhi output (T3S) shares great similarity with Tone2. Despite both tones having rising pitch contours, there exist subtle acoustic differences. However, the difference in underlying representation between T3S and Tone2 remains unclear. By presenting evidence from the alignment pattern between tones and segments, we show that the acoustic differences between Tone2 and T3S arises out of the difference in gestural organizations. The temporal lag between the initiation of the Vowel gesture and that of Tone gesture in T3S is shorter than that in Tone2. We further argue that underlying Tone3 is the source of incomplete neutralization between the Tone2 and T3S. That is, despite the surface similarity, T3S is stored in the mental lexicon as Tone3.

2aSC8. A real-time MRI investigation of anticipatory posturing in prepared responses. Sam Tilsen (Linguist, Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, tilsen@cornell.edu), Pascal Spincemaille (Radiology, Cornell Weill Medical College, New York, NY), Bo Xu (Biomedical Eng., Cornell Univ., New York, NY), Peter Doerschuk (Biomedical Eng., Cornell Univ., Ithaca, NY), Wenming Luh (Human Ecology, Cornell Univ., Ithaca, NY), Robin Karlin, Hao Yi (Linguist, Cornell Univ., Ithaca, NY), and Yi Wang (Biomedical Eng., Cornell Univ., Ithaca, NY)

Speakers can anticipatorily configure their vocal tracts in order to facilitate the production of an upcoming vocal response. We find that this anticipatory articulation results in decoherence of articulatory movements that are otherwise coordinated; moreover, speakers differ in the strategies they employ for response anticipation. Real-time MRI images were acquired from eight native English speakers performing a consonant-vowel response task; the task was embedded in a 2×2 design, which manipulated preparation (whether speakers were informed of the target response prior to a go-signal) and postural constraint (whether the response was preceded by a prolonged vowel). Analyses of pre-response articulatory postures show that all speakers exhibited anticipatory posturing of the tongue root in unconstrained responses. Some exhibited interactions between preparation and constraint, such that anticipatory posturing was more extensive in prepared- vs. unprepared-unconstrained responses. Cross-speaker variation was also observed in anticipatory posturing of the velum: some speakers raised the velum in anticipation of non-nasal responses, while others failed to do so. The results show that models of speech production must be flexible enough to allow for gestures to be executed individually, and that speakers differ in the strategies they employ for response initiation.

2aSC9. An airflow examination of the Czech trills. Ekaterina Komova (East Asian Lang. and Cultures, Columbia Univ., New York, NY) and Phil Howson (The Univ. of Toronto, 644B-60 Harbord St., Toronto, ON M5S3L1, Canada, phil.howson@mail.utoronto.ca)

Previous studies have suggested that there is a difference between the Czech trills /r̝/ and /r/ with respect to the airflow required to produce each trill. This study examines this question using an airflow meter. Five speakers of Czech produced /r̝/ and /r/ in the real words řád "order," pařát "talon," tvař "face," rád "like," paráda "great," and tvar "shape." Airflow data were recorded using Macquirer. The data indicate a higher airflow during the production of /r̝/ compared to /r/. /r̝/ was produced with approximately 3 l/s more than /r/. The increased airflow is necessary to cross the boundary of laminar flow into turbulent flow and supports previous findings that /r̝/ is produced with breathy voice, which facilities trilling during frication. The data also suggests that one of the factors that makes the plain trill /r/ difficult to produce is that the airflow required to produce a sonorous trill is tightly constrained. The boundaries between trill production and the production of frication are only a few l/s apart and thus requires careful management of the laryngeal mechanisms, which control airflow.

2aSC10. Comparison of tidal breathing and reiterant speech breathing using whole body plethysmography. Marguerite Bonadies, Robert H. Withnell, and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 505 W Lava Way, Apt. C, Bloomington, IN 47404, mcbonadi@umail.iu.edu)

Classic research in the field of speech breathing has found differences in the characteristics of breathing patterns between speech respiration and tidal breathing. Though much research has been done on speech breathing mechanisms, relatively little research has been done using the whole body plethysmograph. In this study, we sought to examine differences and similarities between tidal respiration and breathing in reiterant speech using measures obtained through whole-body plethysmography. We hypothesize that there are not significant differences between pulmonary measures in tidal respiration and in speech breathing. This study involves tidal breathing on a spirometer attached to the whole-body plethysmograph followed by reiterant speech using the syllable /da/ while reading the first part of The Rainbow Passage. Experimental measures include compression volumes during both breathing tasks, and absolute lung volumes as determined from the spirometer and calibrated whole-body plethysmograph. These are compared with the pulmonary subdivisions obtained from pulmonary function tests, including vital capacity, functional residual capacity, and total lung volume.

2aSC11. An electroglottography examination of fricative and sonorous segments. Phil Howson (The Univ. of Toronto, 644B-60 Harbord St., Toronto, ON M5S3L1, Canada, phil.howson@mail.utoronto.ca)

It has been previously suggested that fricative production is marked by a longer glottal opening as compared to sonorous segments. The present study uses electroglottography (EGG) and acoustic measurements to test this hypothesis by examining the activity of the vocal cords during the articulation of fricative and sonorant segments of English and Sorbian. An English and a Sorbian speakers' extended individual productions of the phonemes /s, z, ʃ, ʒ, m, n, r, l, a/ and each phoneme in the context #Ca were recorded. The open quotient was calculated using MATLAB. H1-H2 measures were taken at 5% into the vowel following each C and at 50% into the vowel. The results indicate that the glottis is open longer during the production of fricatives than for sonorous segments. Furthermore, the glottis is slightly more open for the production of nasals and liquids than it is for vowels. These results suggest that a longer glottal opening facilitates the increased airflow required to produce frication. This contrasts previous analyses which suggested that frication is primarily achieved through a tightened constriction. While a tighter constriction may be necessary, the increased airflow velocity produced by a longer glottal opening is critical for the production of frication.

2aSC12. SIPMI: Superimposing palatal profile from maxillary impression onto midsagittal articulographic data. Wei-rong Chen and Yueh-chin Chang (Graduate Inst. of Linguist, National Tsing Hua Univ., 2F-5, No. 62, Ln. 408, Zhong-hua Rd., Zhubei City, Hsinchu County-302, Taiwan, waitlong75@gmail.com)

Palatal traces reconstructed by current advanced technologies of real-time mid-sagittal articulatory tracking (e.g., EMA, ultrasound, rtMRI, etc.) are mostly in low-resolution and lack concrete anatomical/orthodontic reference points as firm articulatory landmarks for determining places of articulation. The present study proposes a method of superimposing a physical palatal profile extracted from maxillary impression, onto mid-sagittal articulatory data. The whole palatal/dental profile is first obtained from performing an alginate maxillary impression, and a plaster maxillary mold is made from the impression. Then, the mold is either (1) cut into halves for hand-tracing or (2) 3D-scanned to extract a high resolution mid-sagittal palatal line. The mid-sagittal palatal line made from maxillary mold is further subdivided into articulatory zones, following definitions of articulatory landmarks in the literature (e.g., Catford 1988), by referring to anatomical/orthodontic landmarks imprinted on the mold. Lastly, the high-resolution, articulatorily divided palatal line can be superimposed, by using modified Iterative Closet Point (ICP) algorithm, onto the reconstructed, low-resolution palatal traces in the real-time mid-sagittal articulatory data, so that clearly divided places of articulation on palate can be visualized with articulatory movements. Evaluation results show that both hand-tracing and 3D-scanned palatal profiles yield accurate superimpositions and satisfactory visualizations of place of articulation in our EMA data.

2aSC13. Waveform morphology of pre-speech brain electrical potentials. Silas Smith and Al Yonovitz (Dept. of Commun. Sci. and Disord., The Univ. of Montana, The Univ of Montana, Missoula, MT 59812, silas.smith@umconnect.umt.edu)

The inter- and intra-subject variations of the cortical responses before the initiation of speech were recorded. These evoked potentials were obtained at a sufficient sample rate that both slow negative waves as well as faster neurogenic signals were obtained. The marking point for determining the pre-event time epoch has been an EMG source. The data are typically acquired off-line and later averaged. This research uses a vocal signal as the marking point, and displays in real time the event-related potential. Subjects were 12 males and females. Electrodes were recorded with a silver–silver chloride electrodes positioned at Cz and using the earlobes as reference and ground. A biological preamplifier was used to amplify the weak bioelectric signals 100,000 times. Each time epoch was sampled at 20,000 samples/sec. The frequency response of these amplifiers had a high-pass of 0.1 Hz and a low-pass of 3 kHz. One second of these signals were averaged for 100 trials just prior to the subject initiation of the word "pool." Electrical brain potentials have proven to be extremely useful for diagnosis, treatment, and research in the auditory system, and are expected to be of equal importance for the speech system.

2aSC14. Acoustic correlates of bilingualism: Relating phonetic production to language experience and attitudes. Wai Ling Law (Linguist, Purdue Univ., Beering Hall, 00 North University St., West Lafayette, IN 47907, wlaw@purdue.edu) and Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN)

Researchers tend to quantify degree of bilingualism according to age-related factors such as age of acquisition (Flege, *et al*. 1999, Yeni-Komshian, *et al*. 2000). However, previous research suggests that bilinguals may also show different degrees of accent and patterns of phonetic interaction between their first language (L1) and second language (L2) as a result of factors such as the quantity and quality of L2 input (Flege & Liu, 2001), amount of L1 vs. L2 use (Flege, *et al*. 1999), and attitude toward each language (Moyer, 2007). The goal of this study is to identify gradient properties of speech production that can be related to gradient language experience and attitudes in a bilingual population that is relatively homogeneous in terms of age-related factors. Native Cantonese-English bilinguals living in Hong Kong produced near homophones in both languages under conditions emphasizing one language or the other on different days. Acoustic phonetic variables related to phonological inventory differences between the two languages, including lexical tone/stress, syllable length, nasality, fricative manner and voicing, release of stop, voice onset time, and vowel quality and length, will be quantified and compared to results from a detailed survey of individual speakers' experience and attitudes toward the two languages.

2aSC15. Dialectal variation in affricate place of articulation in Korean. Yoonjnung Kang (Ctr. for French and Linguist, Univ. of Toronto Scarborough, 1265 Military Trail, HW314, Toronto, ON M1C 1A4, Canada, yoonjung.kang@utoronto.ca), Sungwoo Han (Dept. of Korean Lang. and Lit., Inha Univ., Incheon, South Korea), Alexei Kochetov (Dept. of Linguist, Univ. of Toronto, Toronto, ON, Canada), and Eunjong Kong (Dept. of English, Korea Aerosp. Univ., Goyang, South Korea)

The place of articulation (POA) of Korean affricates has been a topic of much discussion in Korean linguistics. The traditional view is that the affricates were dental in the 15th century and then changed to a posterior coronal place in most dialects of Korean but the anterior articulation is retained in major dialects of North Korea, most notably Phyengan and Yukjin. However, recent instrumental studies on Seoul Korean and some impressionistic descriptions of North Korean dialects cast doubt on the validity of this traditional view. Our study examines the POA of /c/ (lenis affricate) and /s/ (anterior fricative) before /a/ in Seoul Korean (26 younger and 32 older speakers) and in two North Korean varieties, as spoken by ethnic Koreans in China (14 Phyengan and 21 Yukjin speakers). The centre of gravity of the frication noise of /c/ and /s/ was examined. The results show that in both North Korean varieties, both sibilants are produced as anterior coronal and comparable in their POA. In Seoul Korean, while the POA contrast shows a significant interaction with age and gender, the affricate is consistently and substantially more posterior than the anterior fricative across all speaker groups. The results support the traditional description.

2aSC16. An articulatory study of high vowels in Mandarin produced by native and non-native speakers. Chenhuei Wu (Dept. of Chinese Lang. and Lit., National Hsinchu Univ. of Education, No. 521, Nanda Rd, Hsinchu 300, Taiwan, chenhueiwu@gmail.com), Weirong Chen (Graduate Inst. of Linguist, National Tsing-hua Univ., Hsinchu, Taiwan), and Chilin Shih (Dept. of Linguist, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

This paper examined the articulatory properties of high vowels [i], [y], and [u] in Mandarin produced by four Taiwanese Mandarin native speakers and four English-speaking Chinese learners (L2 learners) with an Electromagnetic Articulagroph AG500. The articulatory positions of the tongue top (TT), the tongue body (TB), the tongue dorsal (TD), and the lips were investigated. The TT, TB, and TD of [y] produced by the L2 learner were further back than that by the native. In addition, the TD of [y] by the L2 learners was higher than the native. Further comparison found that the tongue positions of [y] was similar to [u] in L2 production. Regarding to the lip positions, the [y] and [u] were more protruded that [i] in the native production, while there is no difference among these three vowels in the L2 production.

The findings suggested that most of the L2 learner were not aware that the lingual target [y] should be very similar to [i] but the lip articulators of [y] are more protruded than [i]. Some L2 learners pronounce [y] more like a diphthong [iu] rather than a monophthong.

2aSC17. Production and perception training of /r l/ with native Japanese speakers. Anna M. Schmidt (School of Speech Path. & Aud., Kent State Univ., A104 MSP, Kent, OH 444242, aschmidt@kent.edu)

Visual feedback with electropalatometry was used to teach accurate /r/ and /l/ to a native Japanese speaker. Perceptual differentiation of the phonemes did not improve. A new perceptual training protocol was developed and tested.

2aSC18. Production of a non-phonemic variant in a second language: Acoustic analysis of Japanese speakers' production of American English flap. Mafuyu Kitahara (School of Law, Waseda Univ., 1-6-1 Nishiwaseda, Shinjuku-ku, Tokyo 1698050, Japan, kitahara@waseda.jp), Keiichi Tajima (Dept. of Psych., Hosei Univ., Tokyo, Japan), and Kiyoko Yoneyama (Dept. of English Lang., Daito Bunka Univ., Tokyo, Japan)

Second-language (L2) learners need to learn the sound system of an L2 so that they can distinguish L2 words. However, it is also instructive to learn non-phonemic, allophonic variations, particularly if learners want to sound native-like. The production of intervocalic /t d/ as an alveolar flap is a prime example of a non-phonemic variation that is salient in American English and presumably noticeable to many L2 learners. Yet, how well such non-phonemic variants are learned by L2 learners is a relatively under-explored subject. In the present study, Japanese learners' production of alveolar flaps was investigated, to clarify how well learners can learn the phonetic environments in which flapping tends to occur, and how L2 experience affects their performance. Native Japanese speakers who had lived in North America for various lengths of time read a list of words and phrases that contained a potentially flappable stop, embedded in a carrier sentence. Preliminary results indicated that the rate of flapping varied considerably across different words and phrases and across speakers. Furthermore, acoustic parameters such as flap closure duration produced by some speakers showed intermediate values between native-like flaps and regular stops, suggesting that flapping is a gradient phenomenon. [Work supported by JSPS.]

2aSC19. A comparison of speaking rate consistency in native and non-native speakers of English. Melissa M. Baese-Berk (Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, mbaesebe@uoregon.edu) and Tuuli Morrill (Linguist, George Mason Univ., Fairfax, VA)

Non-native speech differs from native speech in many ways, including overall longer durations and slower speech rates (Guion *et al.*, 2000). Speaking rate also influences how listeners perceive speech, including perceived fluency of non-native speakers (Munro & Derwing, 1998). However, it is unclear what aspects of non-native speech and speaking rate might influence perceived fluency. It is possible that in addition to differences in mean speaking rate, there may be differences in the consistency of speaking rate within and across utterances. In the current study, we use production data to examine speaking rate in native and non-native speakers of English, and ask whether native and non-native speakers differ in the consistency of their speaking rate across and within utterances. We examined a corpus of read speech, including isolated sentences and longer narrative passages. Specifically, we test whether the overall slower speech rate of non-native speakers is coupled with an inconsistent speech rate that may result in less predictability in the produced speech signal.

2aSC20. Relative distances among English front vowels produced by Korean and American speakers. Byunggon Yang (English Education, Pusan National Univ., 30 Changjundong Keumjunggu, Pusan 609-735, South Korea, bgyang@pusan.ac.kr)

This study examined the relative distances among English front vowels in a message produced by 47 Korean and American speakers from an internet speech archive in order to provide better pronunciation skills for Korean English learners. The Euclidean distances in the vowel space of F1 and F2 were measured among the front vowel pairs. The first vowel pair [i-ɛ] was set as the reference from which the relative distances of the other two vowel pairs were measured in percent in order to compare the vowel sounds among speakers of different vocal tract lengths. Results show that F1 values of the front vowels produced by the Korean and American speakers increased from the high front vowel to the low front vowel with differences among the groups. The Korean speakers generally produced the front vowels with smaller jaw openings than the American speakers did. Second, the relative distance of the high front vowel pair [i-ɪ] showed a significant difference between the Korean and American speakers while that of the low front vowel pair [ɛ-æ] showed a non-significant difference. Finally, the Korean speakers in the higher proficiency level produced the front vowels with higher F1 values than those in the lower proficiency level.

Session 2aUW

Underwater Acoustics: Signal Processing and Ambient Noise

Jorge E. Quijano, Chair
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Contributed Papers

8:00

2aUW1. Moving source localization and tracking based on data. Tsih C. Yang (Inst. of Undersea Technol., National Sun Yat-sen Univ., 70 Lien Hai Rd., Kaohsiung 80404, Taiwan, tsihyang@gmail.com)

Matched field processing (MFP) was introduced sometimes ago for source localization based on the replica field for a hypothesized source location that best matches the acoustic data received on a vertical line array (VLA). A data-based matched-mode source localization method is introduced in this paper for a moving source, using mode wavenumbers and depth functions estimated directly from the data, without requiring any environmental acoustic information and assuming any propagation model to calculate the replica field. The method is in theory free of the environmental mismatch problem since the mode replicas are estimated from the same data used to localize the source. Besides the estimation error due to the approximations made in deriving the data-based algorithms, the method has some inherent drawbacks: (1) it uses a smaller number of modes than theoretically possible, since some modes are not resolved in the measurements, and (2) the depth search is limited to the depth covered by the receivers. Using simulated data, it is found that the performance degradation due to the above approximation/limitation is marginal compared with the original matched-mode source localization method. Certain aspects of the proposed method have previously been tested against data. The key issues are discussed in this paper.

8:15

2aUW2. Simultaneous localization of multiple vocalizing humpback whale calls in an ocean waveguide with a single horizontal array using the array invariant. Zheng Gong, Sunwoong Lee (Mech. Eng., Massachusetts Inst. of Technol., 5-435, 77 Massachusetts Ave., Cambridge, MA 02139, zgong@mit.edu), Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA), and Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

The array invariant method, previously derived for instantaneous range and bearing estimation of a broadband impulsive source in a horizontally stratified ocean waveguide [Lee and Makris, J. Acoust. Soc. Am. 119, 336–351 (2006)], is generalized to instantaneously and simultaneously localize multiple uncorrelated broadband noise sources that not necessarily impulsive in the time domain. In an ideal Pekeris waveguide, we theoretically show that source range and bearing can be instantaneously obtained from beam-time migration lines measured with a horizontal array through range and bearing dependent differences that arise between modal group speeds along the array. We also show that this theory is approximately valid in a horizontally stratified ocean waveguide. A transform, similar to the Radon transform, is employed to enable simultaneous localization of multiple uncorrelated broadband noise sources without ambiguity using the array invariant method. The method is now applied to humpback whale vocalization data from the Gulf of Maine 2006 Experiment for humpback whale ranges up to tens of kilometers, where it is shown that accurate bearing and range estimation of multiple vocalizing humpback whales can be simultaneously made with little computational effort.

8:30

2aUW3. Test for eigenspace stationarity applied to multi-rate adaptive beamformer. Jorge E. Quijano (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, jorgess39@hotmail.com) and Lisa M. Zurk (Elec. and Comput. Eng. Dept., Portland State Univ., Portland, OR)

Array processing in the presence of moving targets is challenging since the number of stationary data snapshots required for estimation of the data covariance are limited. For experimental scenarios that include a combination of fast maneuvering loud interferers and quiet targets, the multi-rate adaptive beamformer (MRABF) can mitigate the effect of non-stationarity. In MRABF, the eigenspace associated to loud interferers is first estimated and removed, followed by application of adaptive beamforming techniques to the remaining, less variable, "target" subspace. Selection of the number of snapshots used for estimation of the interferer eigenspace is crucial in the operation of MRABF, since too few snapshots result in poor eigenspace estimation, while too many snapshots result in leakage of non-stationary interferer effects into the target subspace. In this work an eigenvector-based test for data stationarity, recently developed in the context of very large arrays with snapshot deficiency, is used as a quantitative method to select the optimal number of snapshots for the estimation of the non-stationary eigenspace. The approach is demonstrated with simulated and experimental data from the Shallow Water Array Performance (SWAP) experiment.

8:45

2aUW4. Design of a coprime array for the North Elba sea trial. Vaibhav Chavali, Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., 4307 Ramona Dr., Apt. # H, Fairfax, VA 22030, vchavali@gmu.edu), and John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Dartmouth, MA)

Vaidyanathan and Pal [IEEE Trans. Signal Process. 2011] proposed the use of Coprime Sensor Arrays (CSAs) to sample spatial fields using fewer elements than a Uniform Line Array (ULA) spanning the same aperture. A CSA consists of two interleaved uniform subarrays that are undersampled by coprime factors M and N. The subarrays are processed independently and then their scanned responses are multiplied to obtain a unaliased output. Although the CSA achieves resolution comparable to that of a fully populated ULA, the CSA beampattern has higher sidelobes. Adhikari *et al.* [Proc. ICASSP, 2013] showed that extending the subarrays and applying spatial tapers could reduce CSA sidelobes. This paper considers the problem of designing a CSA for the North Elba Sea Trial described by Gingras [SACLANT Tech. Report, 1994]. The experimental dataset consists of receptions recorded by a 48-element vertical ULA in a shallow water environment for two different source frequencies: 170 Hz and 335 Hz. This paper considers all possible coprime subsamplings for this array and selects the configuration that provides the best tradeoff between number of sensors and performance. Results are shown for both simulated and experimental data. [Work supported by ONR Basic Research Challenge Program.]

9:00

2aUW5. Localization of a high frequency source in a shallow ocean sound channel using frequency-difference matched field processing. Brian Worthmann (Appl. Phys., Univ. of Michigan, 3385 Oakwood St., Ann Arbor, MI 48104, bworthma@umich.edu), H. C. Song (Marine Physical Lab., Scripps Inst. for Oceanogr., Univ. of California - San Diego, La Jolla, CA), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Matched field processing (MFP) is an established technique for locating remote acoustic sources in known environments. Unfortunately, environment-to-propagation model mismatch prevents successful application of MFP in many circumstances, especially those involving high frequency signals. For beamforming applications, this problem was found to be mitigated through the use of a nonlinear array-signal-processing technique called frequency difference beamforming (Abadi *et al*. 2012). Building on that work, this nonlinear technique was extended to MFP, where Bartlett ambiguity surfaces were calculated at frequencies two orders of magnitude lower than the propagated signal, where the detrimental effects of environmental mismatch are much reduced. In the Kauai Acomms MURI 2011 (KAM11) experiment, underwater signals of frequency 11.2 kHz to 32.8 kHz were broadcast 3 km through a 106-m-deep shallow-ocean sound channel and were recorded by a sparse 16-element vertical array. Using the ray-tracing code Bellhop as the propagation model, frequency difference MFP was performed, and some degree of success was found in localizing the high frequency source. In this presentation, the frequency difference MFP technique is explained, and comparisons of this nonlinear MFP technique with conventional Bartlett MFP using both simulations and KAM11 experimental data are provided. [Sponsored by the Office of Naval Research.]

9:15

2aUW6. Transarctic acoustic telemetry. Hee-Chun Song (SIO, UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu), Peter Mikhalvesky (Leidos Holdings, Inc., Arlington, VA), and Arthur Baggeroer (Mech. Eng., MIT, Cambridge, MA)

On April 9 and 13, 1999, two Arctic Climate Observation using Underwater Sound (ACOUS) tomography signals were transmitted from a 20.5-Hz acoustic source moored at the Franz Victoria Strait to an 8-element, 525-m vertical array at ice camp APLIS in the Chukchi Sea at a distance of approximately 2720 km. The transmitted signal was a 20-min long, 255-digit m-sequence that can be treated as a binary-phase shift-keying communication signal with a data rate of 2 bits/s. The almost error-free performance using either spatial diversity (three elements) for a single transmission or temporal diversity (two transmissions) with a single element demonstrates the feasibility of ice-covered trans-Arctic acoustic communications.

9:30

2aUW7. Performance of adaptive multichannel decision-feedback equalization in the simulated underwater acoustic channel. Xueli Sheng, Lina Fan (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin Eng. University Shuisheng Bldg. 803, Nantong St. 145, Harbin, Heilongjiang 150001, China, shengxueli@aliyun.com), Aijun Song, and Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE)

Adaptive multichannel decision feedback equalization [M. Stojanovic, J. Catipovic, and J. G. Proakis, J. Acoust. Soc. Am. 94, 1621–1631 (1993)] is widely adopted to address the severe inter-symbol interference encountered in the underwater acoustic communication channel. In this presentation, its performance will be reported in the simulated communication channel provided by a ray-based acoustic model, for different ocean conditions and source-receiver geometries. The ray model uses the Rayleigh parameter to prescribe the sea surface effects on the acoustic signal. It also supports different types of sediment. The ray model output has been compared with the experimental data and shows comparable results in transmission loss. We will also compare against the performance of multichannel decision feedback equalization supported by existing ray models, for example, BELLHOP.

9:45

2aUW8. Space-time block code with equalization technology for underwater acoustic channels. Chunhui Wang, Xueli Sheng, Lina Fan, Jia Lu, and Weijia Dong (Sci. and Technol. on Underwater Acoust. Lab., College of Underwater Acoust. Engineering, Harbin Eng. Univ., Harbin 150001, China, 740443619@qq.com)

In order to combat the effects of multipath interference and fading for underwater acoustic (UWA) channels, this paper investigates a scheme of the combination of space-time block code (STBC) and equalization technology. STBC is used in this scheme to reduce the effects of fading for the UWA channels, then equalization technology is used in this scheme to mitigate intersymbol interference. The performance of the scheme is analyzed with Alamouti space-time coding for the UWA channels. Our simulations indicate that the combination of STBC and equalization technology provides lower bit error rates.

10:00–10:15 Break

10:15

2aUW9. Robust focusing in time-reversal mirror with a virtual source array. Gi Hoon Byun and Jea Soo Kim (Ocean Eng., Korea Maritime and Ocean Univ., Dongsam 2-dong, Yeongdo-gu, Busan, Korea, Busan, South Korea, knitpia77@gmail.com)

The effectiveness of Time-Reversal (TR) focusing has been demonstrated in various fields of ocean acoustics. In TR focusing, a probe source is required for a coherent acoustic focus at the original probe source location. Recently, the need of a probe source has been partially relaxed by introduction of the concept of a Virtual Source Array (VSA) [S. C. Walker, Philippe Roux, and W. A. Kuperman, J. Acoust. Soc. Am. 125(6), 3828–3834 (2009)]. In this study, Adaptive Time-Reversal Mirror (ATRM) based on multiple constraint method [J. S. Kim, H. C. Song, and W. A. Kuperman, J. Acoust. Soc. Am. 109(5), 1817–1825 (2001)] and Singular Value Decomposition (SVD) method are applied to a VSA for robust focusing. The numerical simulation results are presented and discussed.

10:30

2aUW10. Wind generated ocean noise in deep sea. Fenghua Li and Jingyan Wang (State Key Lab. of Acoust., Inst. of Acoust., CAS, No. 21 Beisihuanxi Rd., Beijing 100190, China, lfh@mail.ioa.ac.cn)

Ocean noise is an important topic in underwater acoustics, which has been paid much attention in last decades. Ocean noise sources may consist of wind, biological sources, ships, earthquakes and so on. This paper discusses measurements of the ocean noise intensity in deep sea during strong wind periods. During the experiment, shipping density is small enough and the wind generated noise is believed to be the dominated effect in the observed frequency range. The analyses of the recoded noise data reveal that the wind generated noise source has a strong dependence on the wind speed and frequency. Based on the data, a wind generated noise source model is presented. [Work supported by National Natural Science Foundation of China, Grant No. 11125420.]

10:45

2aUW11. Ocean ambient noise in the North Atlantic during 1966 and 2013–2014. Ana Sirovic, Sean M. Wiggins, John A. Hildebrand (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr. MC 0205, La Jolla, CA 92093-0205, asirovic@ucsd.edu), and Mark A. McDonald (Whale Acoust., Bellvue, CO)

Low-frequency ocean ambient noise has been increasing in many parts of the worlds' oceans as a result of increased shipping. Calibrated passive acoustic recordings were collected from June 2013 to March 2014 on the south side of Bermuda in the North Atlantic, at a location where ambient noise data were collected in 1966. Monthly and hourly mean power spectra (15–1000 Hz) were calculated, in addition to skewness, kurtosis, and percentile distributions. Average spectrum levels at 40 Hz, representing shipping noise, ranged from 78 to 80 dB re: 1 $\mu Pa^2/Hz$, with a peak in March and minimum in July and August. Values recorded during this recent period were similar to those recorded during 1966. This is different from trends

observed in the Northern Pacific, where ocean ambient noise has been increasing; however, the location of this monitoring site was exposed only to shipping lanes to the south of Bermuda. At frequencies dominated by wind and waves (500 Hz), noise levels ranged from 55 to 66 dB re: 1 μPa^2/Hz, indicating low sea states (2–3) prevailed during the summer, and higher sea states (4–5) during the winter. Seasonally important contribution to ambient sound also came from marine mammals, such as blue and fin whales.

11:00

2aUW12. Adaptive passive fathometer processing of surface-generated noise received by Nested array. Junghun Kim and Jee W. Choi (Marine Sci. and Convergent Technol., Hanyang Univ., 1271 Sa-3-dong, Ansan 426-791, South Korea, Kimjh0927@hanyang.ac.kr)

Recently, a passive fathometer technique using surface-generated ambient noise has been applied to the estimate of bottom profile. This technique performs the beamforming of ambient noise received by a vertical line array to estimate the sub-bottom layer structure as well as water depth. In the previous works, the surface noise signal processing was performed with equally spaced line arrays and the main topic of the research was the comparison of the results estimated using several beamforming techniques. In this talk, the results estimated from the ambient noise received by the Nested vertical line array (called POEMS) which consists of the total 24-elments and four sub-bands are presented. The measurements were made on the eastern coast (East Sea) of Korea. Four kinds of beamforming algorithms are applied to each sub-band and also, nested array processing combining each sub-band signal was performed to obtain the best result. The results are compared to the bottom profiles from the chirp sonar. [This research was supported by the Agency for Defense Development, Korea.]

11:15

2aUW13. Feasibility of low-frequency acoustic thermometry using deep ocean ambient noise in the Atlantic, Pacific, and Indian Oceans. Katherine F. Woolfe and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 672 Brookline St SW, Atlanta, GA 30310, katherine.woolfe@gmail.com)

Previous work has demonstrated the feasibility of passive acoustic thermometry using coherent processing of low frequency ambient noise (1–40 Hz) recorded on triangular hydrophones arrays spaced ~130 km and located in the deep sound channel. These triangular arrays are part of hydroacoustic stations of the International Monitoring System operated by the Comprehensive Nuclear Test Ban Treaty Organization (Woolfe *et al.*, J. Acoust. Soc. Am. 134, 3983). To understand how passive thermometry could potentially be extended to ocean basin scales, we present a comprehensive study of the coherent components of low-frequency ambient noise recorded on five hydroacoustic stations located Atlantic, Pacific, and Indian Oceans. The frequency dependence and seasonal variability of the spatial coherence and directionality of the low-frequency ambient noise were systematically examined at each of the tested site locations. Overall, a dominant coherent component of the low-frequency noise was found to be caused by seasonal ice-breaking events at the poles for test sites that have line-of-sight paths to polar ice. These findings could be used to guide the placement of hydrophone arrays over the globe for future long-range passive acoustic thermometry experiments.

11:30

2aUW14. Ambient noise in the Arctic Ocean measured with a drifting vertical line array. Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, pworcester@ucsd.edu), John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), and John N. Kemp (Woods Hole Oceanographic Inst., Woods Hole, MA)

In mid-April 2013, a Distributed Vertical Line Array (DVLA) with 22 hydrophone modules over a 600-m aperture immediately below the subsurface float was moored near the North Pole. The top ten hydrophones were spaced 14.5 m apart. The distances between the remaining hydrophones increased geometrically with depth. Temperature and salinity were measured by thermistors in the hydrophone modules and ten Sea-Bird MicroCATs. The mooring parted just above the anchor shortly after deployment and subsequently drifted slowly south toward Fram Strait until it was recovered in mid-September 2013. The DVLA recorded low-frequency ambient noise (1953.125 samples per second) for 108 minutes six days per week. Previously reported noise levels in the Arctic are highly variable, with periods of low noise when the wind is low and the ice is stable and periods of high noise associated with pressure ridging. The Arctic is currently undergoing dramatic changes, including reductions in the extent and thickness of the ice cover, the amount of multiyear ice, and the size of the ice keels. The ambient noise data collected as the DVLA drifted will test the hypothesis that these changes result in longer and more frequent periods of low noise conditions than experienced in the past.

Session 2pAA

Architectural Acoustics and Engineering Acoustics: Architectural Acoustics and Audio II

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

Alexander U. Case, Cochair
Sound Recording Technology, University of Massachusetts Lowell, 35 Wilder St., Suite 3, Lowell, MA 01854

Invited Papers

1:00

2pAA1. Defining home recording spaces. Sebastian Otero (Acustic-O, Laurel 14, San Pedro Martir, Tlalpan, Mexico, D.F. 14650, Mexico, sebastian@acustic-o.com)

The idea of home recording has been widely used through out the audio and acoustics community for some time. The effort and investment put into these projects fluctuate in such a wide spectrum that there is no clear way to unify the concept of "home studio," making it difficult for acoustical consultants and clients to reach an understanding on each other project goals. This paper analyses different spaces which vary in terms of privacy, comfort, size, audio quality, budget, type of materials, acoustic treatments, types of projects developed and equipment, but which can all be called "home recording spaces," in order to develop a more specific classification of these environments.

1:20

2pAA2. Vibrato parameterization. James W. Beauchamp (School of Music and Elec. & Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr., Urbana, IL 61801-6824, jwbeauch@illinois.edu)

In an effort to improve the quality of synthetic vibrato many musical instrument tones with vibrato have been analyzed and frequency-vs-time curves have been parameterized in terms of a time-varying offset and a time-varying vibrato depth. Results for variable mean F0 and instrument are presented. Whereas vocal vibrato appears to sweep out the resonance characteristic of the vocal tract, as shown by amplitude-vs-frequency curves for the superposition of a range of harmonics, amplitude-vs-frequency curves for instruments are dominated by hysteresis effects that obscure their interpretation in terms of resonance characteristics. Nevertheless, there is a strong correlation between harmonic amplitude and frequency modulations. An effort is being made to parameterize this effect in order to provide efficient and expressive synthesis of vibrato tones with independent control of vibrato rate and tone duration.

1:40

2pAA3. Get real: Improving acoustic environments in video games. Yuri Lysoivanov (Recording Arts, Tribeca Flashpoint Media Arts Acad., 28 N. Clark St. Ste. 500, Chicago, IL 60602, yuri.lysoivanov@tfa.edu)

As processing power grows the push for realism in video games continues to expand. However, techniques for generating realistic acoustic environments in games have often been limited. Using examples from major releases, this presentation will take a historical perspective on interactive environment design, discuss current methods for modeling acoustic environments in games and suggest specific cases where acoustic expertise can provide an added layer to the interactive experience.

2:00

2pAA4. Applications of telematic mixing consoles in networked audio for musical performance, spatial audio research, and sound installations. David J. Samson and Jonas Braasch (Rensselaer Polytechnic Inst., 1521 6th Ave., Apt. 303, Troy, NY 12180, samsod2@rpi.edu)

In today's technologically driven world, the ability to connect across great distance via Internet Protocol is more important than ever. As the technology evolves, so does the art and science that relies upon it for collaboration and growth. Developing the state of the art system for flexible and efficient routing of networked audio provides a platform for experimental musicians, researchers, and artists to create freely without the restrictions imposed by traditional telepresence. Building on previous development and testing of a telematic mixing console while addressing critical issues with the current platform and current practice, the console allows for the integration of high-quality networked audio into computer assisted virtual environments (CAVE systems), sound and art installations, and other audio driven research projects. Through user study, beta testing, and integration into virtual audio environments, the console has evolved to meet the demand for power and flexibility critical to multi-site collaboration with high-quality networked audio. Areas of concern addressed in development are computational efficiency, system latency, routing architecture, and results of continued user study.

2:20

2pAA5. Twenty years of electronic architecture in the Hilbert Circle Theatre. Paul Scarbrough (Akustiks, 93 North Main St., South Norwalk, CT 06854, pscarbrough@akustiks.com) and Steve Barbar (E-coustic Systems, Belmont, MA)

In 1984, the Circle Theatre underwent a major renovation, transforming the original 3000 + seat venue into a 1780 seat hall with reclaimed internal volume dedicated to a new lobby and an orchestra rehearsal space. In 1996, a LARES acoustic enhancement system replaced the original electronic architecture system, and has been used in every performance since that time. We will discuss details of the renovation, the incorporation of the electronic architecture with other acoustical treatments, system performance over time, and plans for the future.

2:40

2pAA6. Equalization and compression—Friends or foes? Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

These two essential signal processors have overlapping capabilities. Tuning a sound system for any function requires complementary interaction between equalization and compression. The timbral impact of compression is indirect, and can be counterintuitive. A deeper understanding of compression parameters, particularly attack and release, clarifies the connection between compression and tone and makes coordination with equalization more productive.

3:00–3:15 Break

Contributed Papers

3:15

2pAA7. Analysis of room acoustical characteristics by plane wave decomposition using spherical microphone arrays. Jin Yong Jeon, Muhammad Imran, and Hansol Lim (Dept. of Architectural Eng., Hanyang Univ., 17 Haengdang-dong, Seongdong-gu, Seoul, 133791, South Korea, jyjeon@hanyang.ac.kr)

The room acoustical characteristics have been investigated in temporal and spatial structures of room impulse responses (IRs) at different audience positions in real halls. The spherical microphone array of 32-channel is used for measurements process. Specular and diffusive reflections in IRs have been visualized in temporal domain with sound-field decomposition analysis. For plane wave decomposition, the spherical harmonics are used. The beamforming technique is also employed to make directional measurements and for the spatio-temporal characterization of sound field. The directional measurements by beamforming are performed for producing impulse responses for the different directions to characterize the sound. From the estimation of spatial characterization, the reflective surfaces of the hall are indicated as responsible for specular and diffusive reflections.

3:30

2pAA8. Comparing the acoustical nature of a compressed earth block residence to a traditional wood-framed residence. Daniel Butko (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu)

Various lost, misunderstood, or abandoned materials and methods throughout history can serve as viable options in today's impasse of nature and mankind. Similar to the 19th century resurgence of concrete, there is a developing interest in earth as an architectural material capable of dealing with unexpected fluctuations and rising climate changes. Studying the acoustical nature of earthen construction can also serve as a method of application beyond aesthetics and thermal comfort. Innovations using Compressed Earth Block (CEB) have been developed and researched over the past few decades and recently the collaborative focus for a collaborative team of faculty and students at a NAAB accredited College of Architecture, an ABET accredited College of Engineering, and a local chapter of Habitat for Humanity. The multidisciplinary research project resulted in the design and simultaneous construction of both a CEB residence and a conventionally wood-framed version of equal layout, area, volume, apertures, and roof structure on adjacent sites to prove the structural, thermal, economical, and acoustical value of CEB as a viable residential building material. This paper defines acoustical measurements of both residences such as STC, OITC, TL, NC, FFT, frequency responses, and background noise levels prior to occupancy.

3:45

2pAA9. A case study of a high end residential condominium building acoustical design and field performance testing. Erik J. Ryerson and Tom Rafferty (Acoust., Shen Milsom & Wilke, LLC, 2 North Riverside Plaza, Ste. 1460, Chicago, IL 60606, eryerson@smwllc.com)

A high end multi-owner condominium building complex consists of 314 units configured in a central tower with a 39-story central core, as well as 21 and 30 story side towers. A series of project specific acoustical design considerations related to condominium unit horizontal and vertical acoustical separation as well as background noise control for building HVAC systems were developed for the project construction documents and later field tested to confirm conformance with the acoustical design criteria. This paper presents the results of these building wide field tests as well as a discussion of pitfalls encountered during design, construction, and post-construction.

4:00

2pAA10. Innovative ways to make cross laminated timber panels sound-absorptive. Banda Logawa and Murray Hodgson (Mech. Eng., Univ. of Br. Columbia, 2160-2260 West Mall, Vancouver, BC, Canada, logawa_b@yahoo.com)

Cross Laminated Timber (CLT) panels typically consist of several glued layers of wooden boards with orthogonally alternating directions. This cross-laminating process allows CLT panels to be used as load-bearing plate elements similar to concrete slabs. However, they are very sound-reflective, which can lead to concerns about acoustics. Growing interest in applications of CLT panels as building materials in North America has initiated much current research on their acoustical properties. This project is aimed at investigating ways to improve the sound-absorption characteristics of the panels by integrating arrays of Helmholtz-resonator (HR) absorbers into the panels and establishing design guidelines for CLT-HR absorber panels for various room-acoustical applications. To design the new prototype panels, several efforts have been made to measure and analyze the sound-absorption characteristics of the exposed CLT surfaces in multiple buildings in British Columbia, investigate suitable methods and locations to measure both normal and random incidence sound absorption characteristics, study the current manufacturing method of CLT panels, create acoustic models of CLT-HR absorber panels with various shapes and dimensions, and evaluate the sound absorption performance of prototype panels. This paper will report progress on this work.

4:15

2pAA11. Investigate the persistence of sound frequencies Indoor television decors. Mohsen Karami (Dept. of Media Eng., IRIB Univ., No. 8, Dahmetry 4th Alley, Bahar Ave., Kermanshah, Kermanshah 6718839497, Iran, mohsenkarami.ir@gmail.com)

It seems to add to the décor of the studio and make a half-closed spaces and reduce the absorption of waves hitting the studio sound and lasting changes in the frequency of sound energy absorption occurs. To address this issue, the pink noise playback with B&K2260 device, standard time ISO3382, reverberation time in IRIB channels programs twelve decors was measured in various studios. Survey shows values obtained in all the décor, the persistence of high frequencies and this effect occurred regardless of the décor's shape and the studio is.

TUESDAY AFTERNOON, 28 OCTOBER 2014 LINCOLN, 1:25 P.M. TO 5:00 P.M.

Session 2pAB

Animal Bioacoustics: Topics in Animal Bioacoustics II

Cynthia F. Moss, Chair
Psychological and Brain Sci., Johns Hopkins Univ., 3400 N. Charles St., Ames Hall 200B, Baltimore, MD 21218

Chair's Introduction—1:25

Contributed Papers

1:30

2pAB1. Amplitude shifts in the cochlear microphonics of Mongolian gerbils created by noise exposure. Shin Kawai and Hiroshi Riquimaroux (Life and Medical Sci., Doshisha Univ., 1-3 Miyakotani, Tatara, Kyotanabe 610-0321, Japan, hrikimar@mail.doshisha.ac.jp)

The Mongolian gerbil (Meriones unguiculatus) was used to evaluate effects of intense noise exposure on functions of the hair cells. Cochlear microphonics (CM) served an index to show functions of the hair cells. The purpose of this study was to verify which frequency was most damaged by noise exposure and examine relationships between the frequency and the animal's behaviors. We measured growth and recovery of the temporal shifts in amplitude of CM. The CM was recorded from the round window. Test stimuli used were tone bursts (1–45 kHz in half octave step), with duration of 50 ms (5 ms rise/fall times). The subject was exposed to broadband noise (0.5 to 60 kHz) at 90 dB SPL for 5 minutes. Threshold shifts were measured for the testing tone bursts from immediately after the exposure up to 120 minutes after the exposure. Findings showed that reduction in CM amplitude was observed after the noise exposure. Especially, large reduction was produced in a frequency range of 22.4 kHz. However, little reduction was observed around a frequency range of 4 kHz.

1:45

2pAB2. Detection of fish calls by using the small underwater sound recorder. Ikuo Matsuo (Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, matsuo@cs.tohoku-gakuin.ac.jp), Tomohito Imaizumi, and Tmonari Akamatsu (National Res. Inst. of Fisheries Eng., Fisheries Res. Agency, Kamisu, Japan)

Passive acoustic monitoring has been widely used for the survey of marine mammals. This method can be applied for any sounding creatures in the ocean. Many fish, including croaker, grunt, and snapper, produce species-specific low-frequency sounds associated with courtship and spawning behavior in chorus groups. In this paper, the acoustic data accumulated by an autonomous small underwater recorder were used for the sound detection analysis. The recorder was set on the sea floor off the coast of Chosi in Japan (35°40′55″N, 140°49′14″E). The observed signals include not only target fish calls (white croaker) but also calls of another marine life and noises of vessels. We tried to extract the target fish calls out of the sounds. First, recordings were processed by bandpass filter (400–2400 Hz) to eliminate low frequency noise contamination. Second, a low frequency filter applied to extract envelope of the waveform and identified high intensity sound units, which are possibly fish calls. Third, parameter tuning has been conducted to fit the detection of target fish call using absolute received intensity and duration. In this method, 28614 fish calls could be detected from the observed signals during 130 hours Comparing with manually identified fish call, correct detection and false alarm were 0.88 and 0.03, respectively. [This work was supported by CREST, JST.]

2:00

2pAB3. Changes in note order stereotypy during learning in two species of songbird, measured with automatic note classification. Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com)

In addition to mastering the task of performing the individual notes of a song, many songbirds must also learn to produce each note in a stereotyped order. As a bird practices its new song, it may perform thousands of bouts, providing a rich source of information about how note phonology and note type order change during learning. A combination of acoustic landmark descriptions, neural network and hierarchical clustering classifiers, and Markov models of note order made it possible to measure note order stereotypy in two species of songbird. Captive swamp sparrows (*Melospiza melodia*, 11 birds, 92063 notes/bird), and wild tree swallows (*Tachycineta bicolor*, 18 birds, 448 syllables/bird) were recorded song development. The predictability of swamp sparrow note order showed significant increase during the month-long recording period ($F1,162 = 9977$, $p < 0.001$). Note order stereotypy in tree swallows also increased by a significant amount over a month-long field season (Mann-Whitney $V = 12$, p-value < 0.001). Understanding changes in song stereotypy can improve our knowledge of vocal learning, performance, and cultural transmission.

2:15

2pAB4. Plugin architecture for creating algorithms for bioacoustic signal processing software. Christopher A. Marsh, Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720, cmarsh@rohan.sdsu.edu), and David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Newport, OR)

There are several acoustic monitoring software packages that allow for the creation and execution of algorithms that automate detection, classification, and localization (DCL). Algorithms written for one program are generally not portable to other programs, and usually must be written in a specific programming language. We have developed an application programming interface (API) that seeks to resolve these issues by providing a plugin framework for creating algorithms for two acoustic monitoring packages: Ishmael and PAMGuard. This API will allow new detection, classification, and localization algorithms to be written for these programs without requiring knowledge of the monitoring software's source code or inner workings, and lets a single implementation run on either platform. The API also allows users to write DCL algorithms in a wide variety of languages. We hope that this will promote the sharing and reuse of algorithm code. [Funding from ONR.]

2:30

2pAB5. Acoustic detection of migrating gray, humpback, and blue whales in the coastal, northeast Pacific. Regina A. Guazzo, John A. Hildebrand, and Sean M. Wiggins (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9450 Gilman Dr., #80237, La Jolla, CA 92092, rguazzo@ucsd.edu)

Many large cetaceans of suborder Mysticeti make long annual migrations along the California coast. A bottom-mounted hydrophone was deployed in shallow water off the coast of central California and recorded during November 2012 to September 2013. The recording was used to determine the presence of blue whales, humpback whales, and gray whales. Gray whale calls were further analyzed and the number of calls per day and per hour were calculated. It was found that gray whales make their migratory M3 calls at a higher rate than previously observed. There were also more M3 calls recorded at night than during the day. This work will be continued to study the patterns and interactions between species and compared with shore-based survey data.

2:45

2pAB6. Importing acoustic metadata into the Tethys scientific workbench/database. Sean T. Herbert (Marine Physical Lab., Scripps Inst. of Oceanogr., 8237 Lapiz Dr., San Diego, CA 92126, sth.email@gmail.com) and Marie A. Roch (Comput. Sci., San Diego State Univ., San Diego, CA)

Tethys is a temporal-spatial scientific workbench/database created to enable the aggregation and analysis of acoustic metadata from recordings such as animal detections and localizations. Tethys stores data in a specific format and structure, but researchers produce and store data in various formats. Examples of storage formats include spreadsheets, relational databases, or comma-separated value (CSV) text files. Thus, one aspect of the Tethys project has been to focus on providing options to allow data import regardless of the format in which it is stored. Data import can be accomplished in one of two ways. The first is translation, which transforms source data from other formats into the format Tethys uses. Translation does not require any programming, but rather the specification of an import map which associates the researcher's data with Tethys fields. The second method is a framework called Nilus that enables detection and localization algorithms to create Tethys formatted documents directly. Programs can either be designed around Nilus, or be modified to make use of it, which does require some programming. These two methods have been used to successfully import over 4.5 million records into Tethys. [Work funded by NOPP/ONR/BOEM.]

3:00–3:15 Break

3:15

2pAB7. Temporal patterns in detections of sperm whales (Physeter macrocephalus) in the North Pacific Ocean based on long-term passive acoustic monitoring. Karlina Merkens (Protected Species Div., NOAA Pacific Islands Fisheries Sci. Ctr., NMFS/PIFSC/PSD/Karlina Merkens, 1845 Wasp Blvd., Bldg. 176, Honolulu, HI 96818, karlina.merkens@noaa.gov), Anne Simonis (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), and Erin Oleson (Protected Species Div., NOAA Pacific Islands Fisheries Sci. Ctr., Honolulu, HI)

Sperm whales (Physeter macrocephalus), a long-lived, cosmopolitan species, are well suited for long-term studies, and their high amplitude echolocation signals make them ideal for passive acoustic monitoring. NOAA's Pacific Islands Fisheries Science Center has deployed High-frequency Acoustic Recording Packages (200 kHz sampling rate) at 13 deep-water locations across the central and western North Pacific Ocean since 2005. Recordings from all sites were manually analyzed for sperm whale signals, and temporal patterns were examined on multiple scales. There were sperm whale detections at all sites, although the rate of detection varied by location, with the highest rate at Wake Island (15% of samples), and the fewest detections at sites close to the equator ($<1\%$). Only two locations (Saipan and Pearl and Hermes Reef) showed significant seasonal patterns, with more detections in the early spring and summer than in later summer or fall. There were no significant patterns relating to lunar cycles. Analysis of diel variation revealed that sperm whales were detected more during the day and night compared to dawn and dusk at most sites. The variability shown in these results emphasizes the importance of assessing basic biological patterns and variations in the probability of detection before progressing to further analysis, such as density estimation, where the effects of uneven sampling effort could significantly influence results.

3:30

2pAB8. Automatic detection of tropical fish calls recorded on moored acoustic recording platforms. Maxwell B. Kaplan, T. A. Mooney (Biology, Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS50, Woods Hole, MA 02543, mkaplan@whoi.edu), and Jim Partan (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Passive acoustic recording of biological sound production on coral reefs can help identify spatial and temporal differences among reefs; however, the contributions of individual fish calls to overall trends are often overlooked. Given that the diversity of fish call types may be indicative of fish species diversity on a reef, quantifying these call types could be used as a proxy measure for biodiversity. Accordingly, automatic fish call detectors are needed because long acoustic recorders deployments can generate large volumes of data. In this investigation, we report the development and performance of two detectors—an entropy detector, which identifies troughs in entropy (i.e., uneven distribution of entropy across the frequency band of interest, 100–1000 Hz), and an energy detector, which identifies peaks in root mean square sound pressure level. Performance of these algorithms is assessed against a human identification of fish sounds recorded on a coral reef in the US Virgin Islands in 2013. Results indicate that the entropy and energy detectors, respectively, have false positive rates of 9.9% and 9.9% with false negative rates of 28.8% and 31.3%. These detections can be used to cluster calls into types, in order to assess call type diversity at different reefs.

3:45

2pAB9. Social calling behavior in Southeast Alaskan humpback whales (Megaptera novaeangliae): Communication and context. Michelle Fournet (Dept. of Fisheries and Wildlife, Oregon State Univ., 425 SE Bridgeway Ave., Corvallis, OR 97333, mbellalady@gmail.com), Andrew R. Szabo (Alaska Whale Foundation, Petersburg, AK), and David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Newport, OR)

Across their range humpback whales (Megaptera novaeangliae) produce a wide array of vocalizations including song, foraging vocalizations, and a variety of non-song vocalizations known as social calls. This study investigates the social calling behavior of Southeast Alaskan humpback whales from a sample of 299 vocalizations paired with 365 visual surveys collected

over a three-month period on a foraging ground in Frederick Sound in Southeast Alaska. Vocalizations were classified using visual-aural analysis, statistical cluster analyses, and discriminant function analysis. The relationship between vocal behavior and spatial-behavioral context was analyzed using a Poisson log-linear regression (PLL). Preliminary results indicate that some call types were commonly produced while others were rare, and that the greatest variety of calling occurred when whales were clustered. Moreover, calling rates in one vocal class, the pulsed (P) vocal class, were negatively correlated with mean nearest-neighbor distance, indicating that P calling rates increased as animals clustered, suggesting that the use of P calls may be spatially mediated. The data further suggest that vocal behavior varies based on social context, and that vocal behavior trends toward complexity as the potential for social interactions increases. [Work funded by Alaska Whale Foundation and ONR.]

4:00

2pAB10. First measurements of humpback whale song received sound levels recorded from a tagged calf. Jessica Chen, Whitlow W. L. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii at Manoa, 46-007 Lilipuna Rd., Kaneohe, HI 96744, jchen2@hawaii.edu), and Adam A. Pack (Departments of Psych. and Biology, Univ. of Hawaii at Hilo, Hilo, HI)

There is increasing concern over the potential ecological effects from high levels of oceanographic anthropogenic noise on marine mammals. Current US NOAA regulations on received noise levels as well as the Draft Guidance for Assessing the Effect of Anthropogenic Sound on Marine Mammals are based on limited studies conducted on few species. For the regulations to be effective, it is important to first understand what whales hear and their received levels of natural sounds. This novel study presents the measurement of sound pressure levels of humpback whale song received at a humpback whale calf in the wintering area of Maui, Hawaii. This individual was tagged with an Acousonde acoustic and data recording tag and captured vocalizations from a singing male escort associated with the calf and its mother. Although differences in behavioral reaction to anthropogenic versus natural sounds have yet to be quantified, this represents the first known measurement of sound levels that a calf may be exposed to naturally from conspecifics. These levels can also be compared to calculated humpback song source levels. Despite its recovering population, the humpback whale is an endangered species and understanding their acoustic environment is important for continued regulation and protection.

4:15

2pAB11. Seismic airgun surveys and vessel traffic in the Fram Strait and their contribution to the polar soundscape. Sharon L. Nieukirk, Holger Klinck, David K. Mellinger, Karolin Klinck, and Robert P. Dziak (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, sharon.nieukirk@oregonstate.edu)

Low-frequency (<1 kHz) noise associated with human offshore activities has increased dramatically over the last 50 years. Of special interest are areas such as the Arctic where anthropogenic noise levels are relatively low but could change dramatically, as sea ice continues to shrink and trans-polar shipping routes open. In 2009, we began an annual deployment of two calibrated autonomous hydrophones in the Fram Strait to record underwater ambient sound continuously for one year at a sampling rate of 2 kHz. Ambient noise levels were summarized via long-term spectral average plots and reviewed for anthropogenic sources. Vessel traffic data were acquired from the Automatic Identification System (AIS) archive and ship density was estimated by weighting vessel tracklines by vessel length. Background noise levels were dominated by sounds from seismic airguns during spring, summer and fall months; during summer these sounds were recorded in all hours of the day and all days of a month. Ship density in the Fram Strait peaked in late summer and increased every year. Future increases in ship traffic and seismic surveys coincident with melting sea ice will increase ambient noise levels, potentially affecting the numerous species of acoustically active whales using this region.

4:30

2pAB12. Using the dynamics of mouth opening in echolocating bats to predict pulse parameters among individual *Eptesicus fuscus*. Laura N. Kloepper, James A. Simmons (Dept. of Neurosci., Brown Univ., 185 Meeting St. Box GL-N, Brown University Providence, RI 02912, laura_kloepper@brown.edu), and John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Dartmouth, MA)

The big brown bat (Eptesicus fuscus) produces echolocation sounds in its larynx and emits them through its open mouth. Individual mouth-opening cycles last for about 50 ms, with the sound produced in the middle, when the mouth is approaching or reaching maximum gape angle. In previous work, the mouth gape-angle at pulse emission only weakly predicted pulse duration and the terminal frequency of the first-harmonic FM downsweep. In the present study, we investigated whether the dynamics of mouth opening around the time of pulse emission predict additional pulse waveform characteristics. Mouth angle openings for 24 ms before and 24 ms after pulse emission were compared to pulse waveform parameters for three big brown bats performing a target detection task. In general, coupling to the air through the mouth seems less important than laryngeal factors for determining acoustic parameters of the broadcasts. Differences in mouth opening dynamics and pulse parameters among individual bats highlight this relation. [Supported by NSF and ONR.]

4:45

2pAB13. Investigating whistle characteristics of three overlapping populations of false killer whales (*Pseudorca crassidens*) in the Hawaiian Islands. Yvonne M. Barkley, Erin Oleson (NOAA Pacific Islands Fisheries Sci. Ctr., 1845 Wasp Blvd., Bldg. 176, Honolulu, HI 96818, yvonne.barkley@noaa.gov), and Julie N. Oswald (Bio-Waves, Inc., Encinitas, CA)

Three genetically distinct populations of false killer whales Pseudorca crassidens) reside in the Hawaiian Archipelago: two insular populations (one within the main Hawaiian Islands [MHI] and the other within the Northwestern Hawaiian Islands [NWHI]), and a wide-ranging pelagic population with a distribution overlapping the two insular populations. The mechanisms that created and maintain the separation among these populations are unknown. To investigate the distinctiveness of whistles produced by each population, we adapted the Real-time Odontocete Call Classification Algorithm (ROCCA) whistle classifier to classify false killer whale whistles to population based on 54 whistle measurements. 911 total whistles from the three populations were included in the analysis. Results show that the MHI population is vocally distinct, with up to 80% of individual whistles correctly classified. The NWHI and pelagic populations achieved between 48 and 52% correct classification for individual whistles. We evaluated the sensitivity of the classifier to the input whistle measurements to determine which variables are driving the classification results. Understanding how these three populations differ acoustically may improve the efficacy of the classifier and create new acoustic monitoring approaches for a difficult-to-study species.

Session 2pAO

Acoustical Oceanography: General Topics in Acoustical Oceanography

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

Contributed Papers

1:45

2pAO1. Analysis of sound speed fluctuations in the Fram Strait near Greenland during summer 2013. Kaustubha Raghukumar, John A. Colosi (Oceanogr., Naval Postgrad. School, 315B Spanagel Hall, Monterey, CA 93943, kraghuku@nps.edu), and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA)

We analyze sound speed fluctuations in roughly 600 m deep polar waters from a recent experiment. The Thin-ice Arctic Acoustics Window (THAAW) experiment was conducted in the waters of Fram Strait, east of Greenland, during the summer of 2013. A drifting acoustic mooring that incorporated environmental sensors measured temperature and salinity over a period of four months, along a 500 km north-south transect. We examine the relative contributions of salinity-driven polar internal wave activity, and temperature/salinity variability along isopycnal surfaces (spice) on sound speed perturbations in the Arctic. Both internal-wave and spice effects are compared against the more general deep water PhilSea09 measurements. Additionally, internal wave spectra, energies, and modal bandwidth are compared against the well-known Garrett-Munk spectrum. Given the resurgence of interest in polar acoustics, we believe that this analysis will help parameterize sound speed fluctuations in future acoustic propagation models.

2:00

2pAO2. Sound intensity fluctuations due to mode coupling in the presence of nonlinear internal waves in shallow water. Boris Katsnelson (Marine geoSci., Univ. of Haifa, Mt Carmel, Haifa 31905, Israel, katz@phys.vsu.ru), Valery Grogirev (Phys., Voronezh Univ., Voronezh, Russian Federation), and James Lynch (WHOI, Woods Hole, MA)

Intensity fluctuations of the low frequency LFM signals (band 270–330 Hz) were observed in Shallow Water 2006 experiment in the presence of moving train consisting of about seven separate nonlinear internal waves crossing acoustic track at some angle (~ 80°). It is shown that spectrum of the sound intensity fluctuations calculated for time period of radiation (about 7.5 minutes) contains a few peaks, corresponding to predominating frequency ~6.5 cph (and its harmonics) and small peak, corresponding to comparatively high frequency, about 30 cph, which is interpreted by authors as manifestation of horizontal refraction. Values of mentioned frequencies are in accordance with theory of mode coupling and horizontal refraction on moving nonlinear internal waves, developed earlier by authors. [Work was supported by BSF.]

2:15

2pAO3. A comparison of measured and forecast acoustic propagation in a virtual denied area characterized by a heterogeneous data collection asset-network. Yong-Min Jiang and Alberto Alvarez (Res. Dept., NATO-STO-Ctr. for Maritime Res. and Experimentation, Viale San Bartolomeo 400, La Spezia 19126, Italy, jiang@cmre.nato.int)

The fidelity of sonar performance predictions depends on the model used and the quantity and quality of the environmental information that is available. To investigate the impact of the oceanographic information collected by a heterogeneous and near-real time adaptive network of robots in a simulated access denied area, a field experiment (REP13-MED) was conducted by CMRE during August 2013 in an area (70 × 81 km) located offshore La Spezia (Italy), in the Ligurian Sea. The sonar performance assessment makes use of acoustic data recorded by a vertical line array at source—receiver ranges from 0.5 to 30 km. Continuous wave pulses at multiple frequencies (300–600 Hz) were transmitted at two source depths, 25 and 60 meters, at each range. At least 60 pings were collected for each source depth to build up the statistics of the acoustic received level and quantify the measurement uncertainty. A comparison of the acoustic transmission loss measured and predicted using an ocean prediction model (ROMS) assimilating the observed oceanographic data is presented, and the performance of the observational network is evaluated. [Work funded by NATO–Allied Command Transformation]

2:30

2pAO4. Performance assessment of a short hydrophone array for seabed characterization using natural-made ambient noise. Peter L. Nielsen (Res. Dept., STO-CMRE, V.S. Bartolomeo 400, La Spezia 19126, Italy, nielsen@cmre.nato.int), Martin Siderius, and Lanfranco Muzi (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR)

The passive acoustic estimate of seabed properties using natural-made ambient noise received on a glider equipped hydrophone array provides the capability to perform long duration seabed characterization surveys on demand in denied areas. However, short and compact arrays associated with gliders are limited to a few hydrophones and small aperture. Consequently, these arrays exhibit lower resolution of the estimated seabed properties, and the reliability of the environmental estimates may be questionable. The objective of the NATO-STO CMRE sea trial REP14-MED (conducted west of Sardinia, Mediterranean Sea) is to evaluate the performance of a prototype glider array with eight hydrophones in a line and variable hydrophone spacing for seabed characterization using natural-made ambient noise. This prototype array is deployed vertically above the seabed together with a 32-element reference vertical line array. The arrays are moored at different sites with varying sediment properties and stratification. The seabed reflection properties and layering structure at these sites are estimated from ambient noise using both arrays and the results are compared to assess the glider array performance. Synthetic extension of the glider array is performed to enhance resolution of the bottom properties, and the results are compared with these from the longer reference array.

2:45

2pAO5. Species classification of individual fish using the support vector machine. Atsushi Kinjo, Masanori Ito, Ikuo Matsuo (Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai, MIyagi 981-3193, Japan, atsushi.kinjo@gmail.com), Tomohito Imaizumi, and Tomonari Akamatsu (Fisheries Res. Agency, National Res. Inst. of Fisheries Eng., Hasaki, Ibaraki, Japan)

The fish species classification using echo-sounder is important for fisheries. In the case of fish school of mixed species, it is necessary to classify individual fish species by isolating echoes from multiple fish. A broadband signal, which offered the advantage of high range resolution, was applied to detect individual fish for this purpose. The positions of fish were estimated

2p TUE. PM

from the time difference of arrivals by using the split-beam system. The target strength (TS) spectrum of individual fish echo was computed from the isolated echo and the estimated position. In this paper, the Support Vector Machine was introduced to classify fish species by using these TS spectra. In addition, it is well known that the TS spectra are dependent on not only fish species but also fish size. Therefore, it is necessary to classify both fish species and size by using these features. We tried to classify two species and two sizes of schools. Subject species were chub mackerel (*Scomber japonicas*) and Japanese jack mackerel (*Trachurus japonicus*). We calculated the classification rates to limit the training data, frequency bandwidth and tilt angles. It was clarified that the best classification rate was 71.6 %. [This research was supported by JST, CREST.]

3:00–3:15 Break

3:15

2pAO6. Waveform inversion of ambient noise cross-correlation functions in a coastal ocean environment. Xiaoqin Zang, Michael G. Brown, Neil J. Williams (RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149, xzang@rsmas.miami.edu), Oleg A. Godin (ESRL, NOAA, Boulder, CO), Nikolay A. Zabotin, and Liudmila Zabotina (CIRES, Univ. of Colorado, Boulder, CO)

Approximations to Green's functions have been obtained by cross-correlating concurrent records of ambient noise measured on near-bottom instruments at 5 km range in a 100 m deep coastal ocean environment. Inversion of the measured cross-correlation functions presents a challenge as neither ray nor modal arrivals are temporally resolved. We exploit both ray and modal expansion of the wavefield to address the inverse problem using two different parameterizations of the seafloor structure. The inverse problem is investigated by performing an exhaustive search over the relevant parameter space to minimize the integrated squared difference between computed and measured correlation function waveforms. To perform the waveform-based analysis described, it is important that subtle differences between correlation functions and Green's functions are accounted for. [Work supported by NSF and ONR.]

3:30

2pAO7. Application of time reversal to acoustic noise interferometry in shallow water. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru), Oleg Godin (Univ. of Colorado, Boulder, CO), Jixing Qin (State Key Lab, Inst. of Acoust., Beijing, China), Nikolai Zabotin, Liudmila Zabotina (Univ. of Colorado, Boulder, CO), Michael Brown, and Neil Williams (Univ. of Miami, Miami, FL)

Two-point cross-correlations function (CCF) of diffuse acoustic noise approximates the Green's function, which describes deterministic sound propagation between the two measurement points. Similarity between CCFs and Green's functions motivates application to acoustic noise interferometry of the techniques that were originally developed for remote sensing using broadband, coherent compact sources. Here, time reversal is applied to CCFs of the ambient and shipping noise measured in 100 meter-deep water in the Straits of Florida. Noise was recorded continuously for about six days at three points near the seafloor by pairs of hydrophones separated by 5.0, 9.8, and 14.8 km. In numerical simulations, a strong focusing occurs in the vicinity of one hydrophone when the Green's function is back-propagated from the other hydrophone, with the position and strength of the focus being sensitive to density, sound speed, and attenuation coefficient in the bottom. Values of these parameters in the experiment are estimated by optimizing focusing of the back-propagated CCFs. The results are consistent with the values of the seafloor parameters evaluated independently by other means.

3:45

2pAO8. Shear wave inversion in a shallow coastal environment. Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), Jennifer L. Giard (Marine Acoust., Inc., Middletown, RI), James H. Miller, Christopher D. P. Baxter (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Marcia J. Isakson, and Benjamin M. Goldsberry (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Estimation of the shear properties of seafloor sediments in littoral waters is important in modeling the acoustic propagation and predicting the strength of sediments for geotechnical applications. One of the promising approaches to estimate shear speed is by using the dispersion of seismo-acoustic interface (Scholte) waves that travel along the water-sediment boundary. The propagation speed of the Scholte waves is closely related to the shear wave speed over a depth of 1–2 wavelengths into the seabed. A geophone system for the measurement of these interface waves, along with an inversion scheme that inverts the Scholte wave dispersion data for sediment shear speed profiles have been developed. The components of this inversion scheme are a genetic algorithm and a forward model which is based on dynamic stiffness matrix approach. The effects of the assumptions of the forward model on the inversion, particularly the shear wave depth profile, will be explored using a finite element model. The results obtained from a field test conducted in very shallow waters in Davisville, RI, will be presented. These results are compared to historic estimates of shear speed and recently acquired vibracore data. [Work sponsored by ONR, Ocean Acoustics.]

4:00

2pAO9. The effects of pH on acoustic transmission loss in an estuary. James H. Miller (Ocean Eng., Univ. of Rhode Island, URI Bay Campus, 215 South Ferry Rd., Narragansett, RI 02882, miller@egr.uri.edu), Laura Kloepper (Neurosci., Brown Univ., Providence, RI), Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Arthur J. Spivack, Steven D'Hondt, and Cathleen Turner (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI)

Increasing atmospheric CO_2 will cause the ocean to become more acidic with *p*H values predicted to be more than 0.3 units lower over the next 100 years. These lower pH values have the potential to reduce the absorption component of transmission loss associated with dissolved boron. Transmission loss effects have been well studied for deep water where *p*H is relatively stable over time-scales of many years. However, estuarine and coastal *p*H can vary daily or seasonally by about 1 *p*H unit and cause fluctuations in one-way acoustic transmission loss of 2 dB over a range of 10 km at frequencies of 1 kHz or higher. These absorption changes can affect the sound pressure levels received by animals due to identifiable sources such as impact pile driving. In addition, passive and active sonar performance in these estuarine and coastal waters can be affected by these *p*H fluctuations. Absorption changes in these shallow water environments offer a potential laboratory to study their effect on ambient noise due to distributed sources such as shipping and wind. We introduce an inversion technique based on perturbation methods to estimate the depth-dependent *p*H profile from measurements of normal mode attenuation. [Miller and Potty supported by ONR 322OA.]

Session 2pBA

Biomedical Acoustics: Quantitative Ultrasound II

Michael Oelze, Chair
UIUC, 405 N Mathews, Urbana, IL 61801

Contributed Papers

1:30

2pBA1. Receiver operating characteristic analysis for the detectability of malignant breast lesions in acousto-optic transmission ultrasound breast imaging. Jonathan R. Rosenfield (Dept. of Radiology, The Univ. of Chicago, 5316 South Dorchester Ave., Apt. 423, Chicago, IL 60615, jrosenfield@uchicago.edu), Jaswinder S. Sandhu (Santec Systems Inc., Arlington Heights, IL), and Patrick J. La Rivière (Dept. of Radiology, The Univ. of Chicago, Chicago, IL)

Conventional B-mode ultrasound imaging has proven to be a valuable supplement to x-ray mammography for the detection of malignant breast lesions in premenopausal women with high breast density. We have developed a high-resolution transmission ultrasound breast imaging system employing a novel acousto-optic (AO) liquid crystal detector to enable rapid acquisition of full-field breast ultrasound images during routine cancer screening. In this study, a receiver operating characteristic (ROC) analysis was performed to assess the diagnostic utility of our prototype system. Using a comprehensive system model, we simulated the AO transmission ultrasound images expected for a 1-mm malignant lesion contained within a dense breast consisting of 75% normal breast parenchyma and 25% fat tissue. A Gaussian noise model was assumed with SNRs ranging from 0 to 30. For each unique SNR, an ROC curve was constructed and the area under the curve (AUC) was computed to assess the lesion detectability of our system. For SNRs in excess of 10, the analysis revealed AUCs greater than 0.8983, thus demonstrating strong detectability. Our results indicate the potential for using an imaging system of this kind to improve breast cancer screening efforts by reducing the high false negative rate of mammography in premenopausal women.

1:45

2pBA2. 250-MHz quantitative acoustic microscopy for assessing human lymph-node microstructure. Daniel Rohrbach (Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York City, NY 10038, drohrbach@RiversideResearch.org), Emi Saegusa-Beecroft (Dept. of General Surgery, Univ. of Hawaii and Kuakini Medical Ctr., Honolulu, HI), Eugene Yanagihara (Kuakini Medical Ctr., Dept. of Pathol., Honolulu, HI), Junji Machi (Dept. of General Surgery, Univ. of Hawaii and Kuakini Medical Ctr., Honolulu, HI), Ernest J. Feleppa, and Jonathan Mamou (Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY)

We employed quantitative acoustic microscopy (QAM) to measure acoustic properties of tissue microstructure. 32 QAM datasets were acquired from 2, fresh and 11, deparaffinized, 12-μm-thick lymph-node samples obtained from cancer patients. Our custom-built acoustic microscope was equipped with an F-1.16, 250-MHz transducer having a 160-MHz bandwidth to acquire reflected signals from the tissue and a substrate that intimately contacted the tissue. QAM images with a spatial resolution of 7 μm were generated of attenuation (A), speed of sound (SOS), and acoustic impedance (Z). Samples then were stained using hematoxylin and eosin, imaged by light microscopy, and co-registered to QAM images. The spatial resolution and contrast of QAM images were sufficient to distinguish tissue regions consisting of lymphocytes, fat cells and fibrous tissue. Average properties for lymphocyte-dominated tissue were 1552.6 $\pm$ 30 m/s for SOS, 9.53 $\pm$ 3.6 dB/MHz/cm for A, and 1.58 $\pm$ 0.08 Mrayl for Z. Values for Z obtained from fresh samples agreed well with those obtained from 12-μm sections from the same node. Such 2D images provide a basis for developing improved ultrasound-scattering models underlying quantitative ultrasound methods currently used to detect cancerous regions within lymph nodes. [NIH Grant R21EB016117.]

2:00

2pBA3. Detection of sub-micron lipid droplets using transmission-mode attenuation measurements in emulsion phantoms and liver. Wayne Kreider, Ameen Tabatabai (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), and Yak-Nam Wang (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

In liver transplantation, donor liver viability is assessed by the both the amount and type of fat present in the organ. General guidelines dictate that livers with more than 30% fat should not be transplanted; however, a lack of available donor organs has led to the consideration of livers with more fat. As a part of this process, it is desirable to distinguish micro-vesicular fat (< 1 μm droplets) from macro-vesicular fat (~10 μm droplets). A method of evaluating the relative amounts of micro- and macro-fat is proposed based on transmission-mode ultrasound attenuation measurements. For an emulsion of one liquid in another, attenuation comprises both intrinsic losses in each medium and excess attenuation associated with interactions between media. Using an established coupled-phase model, the excess attenuation associated with a monodisperse population of lipid droplets was calculated with physical properties representative of both liver tissue and dairy products. Calculations predict that excess attenuation can exceed intrinsic attenuation and that a well-defined peak in excess attenuation at 1 MHz should occur for droplets around 0.8 μm in diameter. Such predictions are consistent with preliminary transmission-mode measurements in dairy products. [Work supported by NIH grants EB017857, EB007643, EB016118, and T32 DK007779.]

2:15

2pBA4. Using speckle statistics to improve attenuation estimates for cervical assessment. Viksit Kumar and Timothy Bigelow (Mech. Eng., Iowa State Univ., 4112 Lincoln Swing St., Unit 113, Ames, IA 50014, vkumar@iastate.edu)

Quantitative ultrasound parameters like attenuation can be used to observe microchanges in the cervix. To give a better estimate of attenuation we can use speckle properties to classify which attenuation estimates are valid and conform to the theory. For fully developed and only one type of scatterer, Rayleigh distribution models the signal envelope. But in tissues as the number of scatterer type increases and the speckle becomes unresolved Rayleigh model fails. Gamma distribution has been empirically shown to be the best fit among all the distributions. Since more than one scatterer type is present for our work we used a mixture of gamma distributions. EM algorithm was used to find the parameters of the mixture and on basis of that the tissue types were segmented from each other based on the criteria of different scattering properties. Attenuation estimates were then calculated for tissues of the same scattering type only. 67 Women underwent Transvaginal

2p TUE. PM

scan and the attenuation estimates were calculated for them after segregation of tissues on scattering basis. Attenuation was seen to decrease as the time of delivery came closer.

2:30

2pBA5. Using two-dimensional impedance maps to study weak scattering in isotropic random media. Adam Luchies and Michael Oelze (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Matthews Ave, Urbana, IL 61801, luchies1@illinois.edu)

An impedance map (ZM) is a computational tool for studying weak scattering in soft tissues. Currently, three-dimensional (3D) ZMs are created from a series of adjacent histological slides that have been stained to emphasize acoustic scattering structures. The 3D power spectrum of the 3DZM may be related to quantitative ultrasound parameters such as the backscatter coefficient. However, constructing 3DZMs is expensive, both in terms of computational time and financial cost. Therefore, the objective of this study was to investigate using two-dimensional (2D) ZMs to estimate 3D power spectra. To estimate the 3D power spectrum using 2DZMs, the autocorrelation of 2DZMs extracted from a volume were estimated and averaged. This autocorrelation estimate was substituted into the 3D Fourier transform that assumes radial symmetry to estimate the 3D power spectrum. Simulations were conducted on sparse collections of spheres and ellipsoids to validate the proposed method. Using a single slice that intersected approximately 75 particles, a mean absolute error was achieved of 1.1 dB and 1.5 dB for sphere and ellipsoidal particles, respectively. The results from the simulations suggest that 2DZMs can provide accurate estimates of the power spectrum and are a feasible alternative to the 3DZM approach.

2:45

2pBA6. Backscatter coefficient estimation using tapers with gaps. Adam Luchies and Michael Oelze (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Matthews Ave., Urbana, IL 61801, luchies1@illinois.edu)

When using the backscatter coefficient (BSC) to estimate quantitative ultrasound (QUS) parameters such as the effective scatterer diameter (ESD) and the effective acoustic concentration (EAC), it is necessary to assume that the interrogated medium contains diffuse scatterers. Structures that invalidate this assumption can significantly affect the estimated BSC parameters in terms of increased bias and variance and decrease performance when classifying disease. In this work, a method was developed to mitigate the effects of non-diffuse echoes, while preserving as much signal as possible for obtaining diffuse scatterer property estimates. Specially designed tapers with gaps that avoid signal truncation were utilized for this purpose. Experiments from physical phantoms were used to evaluate the effectiveness of the proposed BSC estimation methods. The mean squared error (MSE) for BSC between measured and theoretical had an average value of approximately 1.0 and 0.2 when using a Hanning taper and PR taper, respectively, with six gaps. The BSC error due to amplitude bias was smallest for PR tapers with time-bandwidth product $N\omega = 1$. The BSC error due to shape bias was smallest for PR tapers with $N\omega = 4$. These results suggest using different taper types for estimating ESD versus EAC.

3:00

2pBA7. Application of the polydisperse structure function to the characterization of solid tumors in mice. Aiguo Han and William D. O'Brien (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., Urbana, IL 61801, han51@uiuc.edu)

A polydisperse structure function model has been developed for modeling ultrasonic scattering from dense scattering media. The polydisperse structure function is incorporated to a fluid-filled sphere scattering model to model the backscattering coefficient (BSC) of solid tumors in mice. Two types of tumors were studied: a mouse sarcoma (Englebreth-Holm-Swarm [EHS]) and a mouse carcinoma (4T1). The two kinds of tumors had significantly different microstructures. The carcinoma had a uniform distribution of carcinoma cells. The sarcoma had cells arranged in groups usually containing less than 20 cells per group, causing an increased scatterer size and size distribution. Excised tumors (13 EHS samples and 15 4T1 samples) were scanned using single-element transducers covering the frequency range 11–105 MHz. The BSC was estimated using a planar reference technique. The model was fit to the experimental BSC using a least-square fit. The mean scatterer radius and the Schulz width factor (which characterizes the width of the scatterer size distribution) were estimated. The results showed significantly higher scatterer size estimates and wider scatterer size distribution estimates for EHS than for 4T1, consistent with the observed difference in microstructure of the two types of tumors. [Work supported by NIH CA111289.]

3:15–3:30 Break

3:30

2pBA8. Experimental comparison of methods for measuring backscatter coefficient using single element transducers. Timothy Stiles and Andrew Selep (Phys., Monmouth College, 700 E Broadway Ave., Monmouth, IL 61462, tstiles@monmouthcollege.edu)

The backscatter coefficient (BSC) has promise as a diagnostic aid. However, measurements of the BSC of soft-tissue mimicking materials have proven difficult; results on the same samples by various laboratories have up to two orders of magnitude difference. This study compares methods of data analysis using data acquired from the same samples using single element transducers, with a frequency range of 1 to 20 MHz and pressure focusing gains between 5 and 60. The samples consist of various concentrations of milk in agar with scattering from glass microspheres. Each method utilizes a reference spectrum from a planar reflector but differ in the diffraction and attenuation correction algorithms. Results from four methods of diffraction correction and three methods of attenuation correction are compared to each other and to theoretical predictions. Diffraction correction varies from no correction to numerical integration of the beam throughout the data acquisition region. Attenuation correction varies from limited correction for the attenuation up to the start of the echo acquisition window, to correcting for attenuation within a numerical integration of the beam profile. Results indicate the best agreements with theory are the methods that utilize the numerical integration of the beam profile.

3:45

2pBA9. Numerical simulations of ultrasound-pulmonary capillary interaction. Brandon Patterson (Mech. Eng., Univ. of Michigan, 626 Spring St., Apt. #1, Ann Arbor, MI 48103-3200, awesome@umich.edu), Douglas L. Miller (Radiology, Univ. of Michigan, Ann Arbor, MI), David R. Dowling, and Eric Johnsen (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Although lung hemorrhage (LH) remains the only bioeffect of non-contrast, diagnostic ultrasound (DUS) proven to occur in mammals, a fundamental understanding of DUS-induced LH remains lacking. We hypothesize that the fragile capillary beds near the lungs surface may rupture as a result of ultrasound-induced strains and viscous stresses. We perform simulations of DUS waves propagating in tissue (modeled as water) and impinging on a planar lung surface (modeled as air) with hemispherical divots representing individual capillaries (modeled as water). Experimental ultrasound pulse waveforms of frequencies 1.5–7.5 MHz are used for the simulation. A high-order accurate discontinuity-capturing scheme solves the two-dimensional, compressible Navier-Stokes equations to obtain velocities, pressures, stresses, strains, and displacements in the entire domain. The mechanics of the capillaries are studied for a range of US frequencies and amplitudes. Preliminary results indicate a strong dependence of the total strain on the capillary size relative to the wavelength.

4:00

2pBA10. Acoustic radiation force due to nonaxisymmetric sound beams incident on spherical viscoelastic scatterers in tissue. Benjamin C. Treweek, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, btreweek@utexas.edu)

The theory for acoustic radiation force on a viscoelastic sphere of arbitrary size in tissue was extended recently to account for nonaxisymmetric incident fields [Ilinskii *et al.*, POMA **19**, 045004 (2013)]. A spherical harmonic expansion was used to describe the incident field. This work was specialized at the spring 2014 ASA meeting to focused axisymmetric sound beams with various focal spot sizes and a scatterer located at the focus. The emphasis of the present contribution is nonaxisymmetric fields, either through moving the scatterer off the axis of an axisymmetric beam or through explicitly defining a nonaxisymmetric beam. This is accomplished via angular spectrum decomposition of the incident field, spherical wave expansions of the resulting plane waves about the center of the scatterer, Wigner D-matrix transformations to express these spherical waves in a coordinate system with the polar axis aligned with the desired radiation force component, and finally integration over solid angle to obtain spherical wave amplitudes as required in the theory. Various scatterer sizes and positions relative to the focus are considered, and the effects of changing properties of both the scatterer and the surrounding tissue are examined. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

4:15

2pBA11. Convergence of Green's function-based shear wave simulations in models of elastic and viscoelastic soft tissue. Yiqun Yang (Dept. of Elec. and Comput. Eng., Michigan State Univ., Michigan State University, East Lansing, MI, yiqunyang.nju@gmail.com), Matthew Urban (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN), and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Green's functions effectively simulate shear waves produced by an applied acoustic radiation force in elastic and viscoelastic soft tissue. In an effort to determine the optimal parameters for these simulations, the convergence of Green's function-based calculations is evaluated for realistic spatial distributions of the initial radiation force "push." The input to these calculations is generated by FOCUS, the "Fast Object-oriented C++ Ultrasound Simulator," which computes the approximate intensity fields generated by a Phillips L7-4 ultrasound transducer array for both focused and unfocused beams. The radiation force in the simulation model, which is proportional to the simulated intensity, is applied for 200 μs, and the resulting displacements are calculated with the Green's function model. Simulation results indicate that, for elastic media, convergence is achieved when the intensity field is sampled at roughly one-tenth of the wavelength of the compressional component that delivers the radiation force "push." Aliasing and oscillation artifacts are observed in the model for an elastic medium at lower sampling rates. For viscoelastic media, spatial sampling rates as low as two samples per compressional wavelength are sufficient due to the low-pass filtering effects of the viscoelastic medium. [Supported in part by NIH Grants R01EB012079 and R01DK092255.]

4:30

2pBA12. Quantifying mechanical heterogeneity of breast tumors using quantitative ultrasound elastography. Tengxiao Liu (Dept. of Mech., Aerosp. and Nuclear Eng., Rensselaer Polytechnic Inst., Troy, NY), Olalekan A. Babaniyi (Mech. Eng., Boston Univ., Boston, MA), Timothy J. Hall (Medical Phys., Univ. of Wisconsin, Wisconsin, WI), Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu), and Assad A. Oberai (Dept. of Mech., Aerosp. and Nuclear Eng., Rensselaer Polytechnic Inst., Troy, NY)

Heterogeneity is a hallmark of cancer whether one considers the genotype of cancerous cells, the composition of their microenvironment, the distribution of blood and lymphatic microvasculature, or the spatial distribution of the desmoplastic reaction. It is logical to expect that this heterogeneity in tumor microenvironment will lead to spatial heterogeneity in its mechanical properties. In this study we seek to quantify the mechanical heterogeneity within malignant and benign tumors using ultrasound based elasticity imaging. By creating *in-vivo* elastic modulus images for ten human subjects with breast tumors, we show that Young's modulus distribution in cancerous breast tumors is more heterogeneous when compared with tumors that are not malignant, and that this signature may be used to distinguish malignant breast tumors. Our results complement the view of cancer as a heterogeneous disease by demonstrating that mechanical properties within cancerous tumors are also spatially heterogeneous. [Work supported by NIH, NSF.]

4:45

2pBA13. Convergent field elastography. Michael D. Gray, James S. Martin, and Peter H. Rogers (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332-0405, michael.gray@me.gatech.edu)

An ultrasound-based system for non-invasive estimation of soft tissue shear modulus will be presented. The system uses a nested pair of transducers to provide force generation and motion measurement capabilities. The outer annular element produces a ring-like ultrasonic pressure field distribution. This in turn produces a ring-like force distribution in soft tissue, whose response is primarily observable as a shear wave field. A second ultrasonic transducer nested inside the annular element monitors the portion of the shear field that converges to the center of the force distribution pattern. Propagation speed is estimated from shear displacement phase changes resulting from dilation of the forcing radius. Forcing beams are modulated in order to establish shear speed frequency dependence. Prototype system data will be presented for depths of 10–14 cm in a tissue phantom, using drive parameters within diagnostic ultrasound safety limits. [Work supported by ONR and the Neely Chair in Mechanical Engineering, Georgia Institute of Technology.]

5:00

2pBA14. Differentiation of benign and malignant breast lesions using Comb-Push Ultrasound Shear Elastography. Max Denis, Mohammad Mehmohammadi (Physiol. and Biomedical Eng., Mayo Clinic, 200 First St. SW, Rochester, MN 55905, denis.max@mayo.edu), Duane Meixner, Robert Fazzio (Radiology-Diagnostic, Mayo Clinic, Rochester, MN), Shigao Chen, Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN), and Azra Alizad (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, Missouri)

In this work, the results from our Comb Push Ultrasound Shear Elastography (CUSE) assessment of suspicious breast lesions are presented. The elasticity value of the breast lesions are correlated to histopathological findings to evaluate their diagnostic value in differentiating between malignant and benign breast lesions. A total of 44 patients diagnosed with suspicious breast lesions were evaluated using CUSE prior to biopsy. All patient study procedures were conducted according to the protocol approved by Mayo Clinic Institutional Review Board (IRB). Our cohort consisted of 27 malignant and 17 benign breast lesions. The results indicate an increase in shear wave velocity in both benign and malignant lesions compared to normal breast tissue. Furthermore, the Young's modulus is significantly higher in malignant lesions. An optimal cut-off value of the Young's modulus $\geq$ 80 kPa was observed for the receiver operating characteristic (ROC) curve. This is concordant with the published cut-off values of elasticity for suspicious breast lesions. [This work is supported in part by the grant 3R01CA148994-04S1 and 5R01CA148994-04 from NIH.]

5:15

2pBA15. Comparison between diffuse infrared and acoustic transmission over the human skull. Qi Wang, Namratha Reganti, Yutoku Yoshioka, Mark Howell, and Gregory T. Clement (BME, LRI, Cleveland Clinic, 9500 Euclid Ave., Cleveland, OH 44195, qiqiwang83@gmail.com)

Skull-induced distortion and attenuation present a challenge to both transcranial imaging and therapy. Whereas therapeutic procedures have been successful in offsetting aberration using from prior CTs, this approach impractical for imaging. In effort to provide a simplified means for aberration correction, we have been investigating the use of diffuse infrared light as an indicator of acoustic properties. Infrared wavelengths were specifically selected for tissue penetration; however this preliminary study was performed through bone alone via a transmission mode to facilitate comparison with acoustic measurements. The inner surface of a half human skull, cut along the sagittal midline, was illuminated using an infrared heat lamp and images of the outer surface were acquired with an IR-sensitive camera. A range of source angles were acquired and averaged to eliminate source bias. Acoustic measurement were likewise obtained over the surface with a source (1 MHz, 12.7 mm-diam) oriented parallel to the skull surface and hydrophone receiver (1 mm PVDF). Preliminary results reveal a positive correlation between sound speed and optical intensity, whereas poor correlation is observed between acoustic amplitude and optical intensity. [Work funded under NIH R01EB014296.]

5:30

2pBA16. A computerized tomography system for transcranial ultrasound imaging. Sai Chun Tang (Dept. of Radiology, Harvard Med. School, 221 Longwood Ave., Rm. 521, Boston, MA 02115, sct@bwh.harvard.edu) and Gregory T. Clement (Dept. of Biomedical Eng., Cleveland Clinic, Cleveland, OH)

Hardware for tomographic imaging presents both challenge and opportunity for simplification when compared with traditional pulse-echo imaging systems. Specifically, point diffraction tomography does not require simultaneous powering of elements, in theory allowing just a single transmit channel and a single receive channel to be coupled with a switching or multiplexing network. In our ongoing work on transcranial imaging, we have developed a 512-channel system designed to transmit and/or receive a high voltage signal from/to arbitrary elements of an imaging array. The overall design follows a hierarchy of modules including a software interface, microcontroller, pulse generator, pulse amplifier, high-voltage power converter, switching mother board, switching daughter board, receiver amplifier, analog-to-digital converter, peak detector, memory, and USB communication. Two pulse amplifiers are included, each capable producing up to 400 Vpp via power MOSFETS. Switching is based around mechanical relays that allow passage of 200 V, while still achieving switching times of under 2 ms, with an operating frequency ranging from below 100 kHz to 10 MHz. The system is demonstrated through ex vivo human skulls using 1 MHz transducers. The overall system design is applicable to planned human studies in transcranial image acquisition, and may have additional tomographic applications for other materials necessitating a high signal output. [Work was supported by NIH R01 EB014296.]

TUESDAY AFTERNOON, 28 OCTOBER 2014 — INDIANA C/D, 2:45 P.M. TO 3:30 P.M.

Session 2pEDa

Education in Acoustics: General Topics in Education in Acoustics

Uwe J. Hansen, Chair

Chemistry& Physics, Indiana State University, 64 Heritage Dr., Terre Haute, IN 47803-2374

Contributed Papers

2:45

2pEDa1. @acousticsorg: The launch of the Acoustics Today twitter feed. Laura N. Kloepper (Dept. of Neurosci., Brown Univ., 185 Meeting St. Box GL-N, Providence, RI 02912, laura_kloepper@brown.edu) and Daniel Farrell (Web Development office, Acoust. Society of America, Melville, NY)

Acoustics Today has recently launched our twitter feed, @acousticsorg. Come learn how we plan to spread the mission of Acoustics Today, promote the science of acoustics, and connect with acousticians worldwide! We will also discuss proposed upcoming social media initiatives and how you, an ASA member, can help contribute. This presentation will include an extended question period in order to gather feedback on how Acoustics Today can become more involved with social media.

3:00

2pEDa2. Using Twitter for teaching. William Slaton (Phys. & Astronomy, The Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72034, wvslaton@uca.edu)

The social media microblogging platform, Twitter, is an ideal avenue to learn about new science in the field of acoustics as well as to share that new-found information with students. As a user discovers a network of science bloggers and journalists to follow the amount of science uncovered grows. Conversations between science writers and scientists themselves enhance this learning opportunity. Several examples of using twitter for teaching will be presented.

3:15

2pEDa3. Unconventional opportunities to recruit future science, technology, engineering, and math scholars. Roger M. Logan (Teledyne, 12338 Westella, Houston, TX 77077, rogermlogan@sbcglobal.net)

Pop culture conventions provide interesting and unique opportunities to inspire the next generation of STEM contributors. Literary, comic, and anime are a few example of this type of event. This presentation will provide insights into these venues as well as how to get involved and help communicate that careers in STEM can be fun and rewarding.

Session 2pEDb

Education in Acoustics: Take 5's

Uwe J. Hansen, Chair
Chemistry& Physics, Indiana State University, 64 Heritage Dr., Terre Haute, IN 47803-2374

For a Take-Five session no abstract is required. We invite you to bring your favorite acoustics teaching ideas. Choose from the following: short demonstrations, teaching devices, or videos. The intent is to share teaching ideas with your colleagues. If possible, bring a brief, descriptive handout with enough copies for distribution. Spontaneous inspirations are also welcome. You sign up at the door for a five-minute slot before the session starts. If you have more than one demo, sign-up for two consecutive slots.

Session 2pID

Interdisciplinary: Centennial Tribute to Leo Beranek's Contributions in Acoustics

William J. Cavanaugh, Cochair
Cavanaugh Tocci Assoc. Inc., 3 Merifield Ln., Natick, MA 01760-5520

Carl Rosenberg, Cochair
Acentech Incorporated, 33 Moulton Street, Cambridge, MA 02138

Chair's Introduction—1:55

Invited Papers

2:00

2pID1. Leo Beranek's role in the Acoustical Society of America. Charles E. Schmid (10677 Manitou Pk. Blvd., Bainbridge Island, WA 98110, cechmid@att.net)

Leo Beranek received the first 75th anniversary certificate issued by the Acoustical Society of America commemorating his long-time association with the Society at the joint ASA/ICA meeting in Montreal in 2013. Both the Society and Leo have derived mutual benefits from this long and fruitful association. Leo has held many important roles as leader in the ASA. He served as vice president (1949–1950), president (1954–1955), Chair of the Z24 Standards Committee (1950–1953), meeting organizer (he was an integral part of the Society's 25th, 50th, and 75th Anniversary meetings), associate editor (1950–1959), author of three books sold via ASA, publisher of 75 peer-reviewed JASA papers, and presented countless papers at ASA meetings. Much of his work has been recognized by the Society which presented him with the R. Bruce Lindsay Award (1944), the Wallace Clement Sabine Award (1961), the Gold Medal (1975), and an Honorary Fellowship (1994). He has participated in the Acoustical Society Foundation and donated generously to it. He has been an inspiration for younger Society members (which include all of us on this occasion celebrating his 100th birthday).

2:15

2pID2. Leo Beranek's contributions to noise control. George C. Maling (INCE FOUNDATION, 60 High Head Rd., Harpswell, ME 04079, INCEUSA@aol.com) and William W. Lang (INCE FOUNDATION, Poughkeepsie, New York)

Leo Beranek has made contributions to noise control for many years, beginning with projects during World War II when he was a Harvard University. Later, at MIT, he taught a course (6.35) which included noise control, and ran MIT summer courses on the subject. His book, *Noise Reduction*, was published during that time. Additional books followed. Noise control became an important part of the consulting work at Bolt Beranek and Newman. Two projects are of particular interest: The efforts to silence a wind tunnel in Cleveland, Ohio, and the differences in noise emissions and perception as the country entered the jet age. Leo was one of the founders of the Institute of Noise Control Engineering, and served as its charter president. Much of the success of the Institute is due to his early leadership. He has also played an important role in noise policy, beginning in the late 1960s and, in particular, with the passage of the Noise Control Act of 1972. This work continued into the 1990s with the formation of the "Peabody Group," and cooperation with the National Academy of Engineering in the formation of noise policy.

2:30

2pID3. Beranek's porous material model: Inspiration for advanced material analysis and design. Cameron J. Fackler and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, facklc@rpi.edu)

In 1942, Leo Beranek presented a model for predicting the acoustic properties of porous materials [J. Acoust. Soc. Am. 13, 248 (1942)]. Since then, research into many types of porous materials has grown into a broad field. In addition to Beranek's model, many other models for predicting the acoustic properties of porous materials in terms of key physical material parameters have been developed. Following a brief historical review, this work concentrates on studying porous materials and microperforated panels—pioneered by one of Beranek's early friends and fellow students, Dah-You Maa. Utilizing equivalent fluid models, porous material and microperforated panel theories have recently been unified. In this work, the Bayesian inference framework is applied to single- and multilayered porous and microperforated materials. Bayesian model selection and parameter estimation are used to guide the analysis and design of innovative multilayer acoustic absorbers.

2:45

2pID4. Technology, business, and civic visionary. David Walden (retired from BBN, 12 Linden Rd., East Sandwich 02537, dave@walden-family.com)

In high school and college, Leo Beranek was already developing the traits of an entrepreneur. At Bolt Beranek and Newman he built a culture of innovation. He and his co-founders also pursued a policy of looking for financial returns, via diversification and exploitation of intellectual property, beyond their initial acoustics-based professional services business. In particular, in 1956–1957 Leo recruited J.C.R. Licklider to help BBN move into the domain of computers. In time, information sciences and computing became as significant a business for BBN as acoustics. While BBN did innovative work in many areas of computing, perhaps the most visible area was with the technology that became the Internet. In 1969, Leo left day-to-day management of BBN, although he remained associated with the company for more years. Beyond BBN, Leo worked, often in a leadership role, with a variety of institutions to improve civic life and culture around Boston.

3:00

2pID5. Leo Beranek and concert hall acoustics. Benjamin Markham (Acentech Inc, 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

Dr. Leo Beranek's pioneering concert hall research and project work has left an indelible impression on the study and practice of concert hall design. Working as both scientist and practitioner simultaneously for most of his 60 + years in the industry, his accomplishments include dozens of published papers on concert hall acoustics, several seminal books on the subject, and consulting credit for numerous important performance spaces. This paper will briefly outline a few of his key contributions to the field of concert hall acoustics (including his work regarding audience absorption, the loudness parameter G, the system of concert hall metrics and ratings that he developed, and other contributions), his project work (including the Tanglewood shed, Philharmonic Hall, Tokyo Opera City concert hall, and others), and his role as an inspiration for other leaders in the field. His work serves as the basis, the framework, the inspiration, or the jumping-off point for a great deal of current concert hall research, as evidenced by the extraordinarily high frequency with which his work is cited; this paper will conclude with some brief remarks on the future of concert hall research that will build on Dr. Beranek's extraordinary career.

3:15

2pID6. Concert hall acoustics: Recent research. Leo L. Beranek (Retired, 10 Longwood Dr., Westwood, MA 02090, beranekleo@ieee.org)

Recent research on concert hall acoustics is reviewed. Discussed are (1) ten top quality halls acoustically; (2) listeners acoustical preferences; (3) how musical dynamics are enhanced by hall shape; (4) effect of seat upholstering on sound strength and hall dimensions; (5) recommended minimum and maximum hall dimensions and audience capacities in shoebox, surround, and fan shaped halls.

3:30

2pID7. Themes of thoughts and thoughtfulness. Carl Rosenberg (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, crosenberg@acentech.com) and William J. Cavanaugh (Cavanaugh/Tocci, Sudbury, MA)

In preparing and compiling the background for the issue of Acoustics Today on Leo Beranek to commemorate his 100th birthday, there were some consistent themes of Leo's work and contribution to colleagues and scholars with whom he worked. This was particularly evident in the many "side-bars" solicited from over three dozen friends and colleagues. The authors discuss these patterns and share insights on the manner in which Leo was most influential. There will be opportunities for audience participants to share their thoughts and birthday greetings with Leo.

3:45–4:15 Panel Discussion

4:15–5:00 Celebration

Session 2pMU

Musical Acoustics: Synchronization Models in Musical Acoustics and Psychology

Rolf Bader, Chair
Institute of Musicology, University of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany

Invited Papers

1:00

2pMU1. Models and findings of synchronization in musical acoustics and music psychology. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

Synchronization is a crucial mechanism in music tone production and perception. With wind instruments, the overtone series of notes synchronize to nearly perfect harmonic relations due to nonlinear effects and turbulence at the driving mechanism although the overblown pitches of flutes or horns may differ considerably from such a simple harmonic relation. Organ pipes close to each other synchronize in pitch by interaction of the sound pressures. With violins, the sawtooth motion appears because of a synchronization of the stick/slip interaction with the string length. All these models are complex systems also showing bifurcations in terms of multiphonics, biphonation or subharmonics. On the subjects perception and music production side models of synchronization, like the free-energy principle modeling perception by minimizing surprise and adaptation to physical parameters of sound production, neural nets of timbre, tone, or rhythm perception or synergetic models of rhythm production are generally suited much better to model music perception than simplified linear models.

1:20

2pMU2. One glottal airflow—Two vocal folds. Ingo R. Titze (National Ctr. for Voice and Speech, Univ. of Utah, 156 South Main St., Ste. 320, Salt Lake City, UT 84101-3306, ingo.titze@utah.edu) and Ingo R. Titze (Dept. of Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA)

Vocalization for speech and singing involves self-sustained oscillation between a stream of air and a pair of vocal folds. Each vocal fold has its own set of natural frequencies (modes of vibration) governed by the viscoelastic properties of tissue layers and their boundary conditions. Due to asymmetry, the modes of left and right vocal folds are not always synchronized. The common airflow between them can entrain the modes, but not always in a 1:1 ratio. Examples of bifurcations are given for human and animal vocalization, as well as from computer simulation. Vocal artists may use desynchronization for special vocal effects. Surgeons who repair vocal folds make decisions about the probability of regaining synchronization when one vocal fold is injured. Complete desynchronization, allowing only one vocal fold to oscillate, may be a better strategy in some cases then attempting to achieve symmetry.

1:40

2pMU3. Synchronization of organ pipes—Experimental facts and theory. Markus W. Abel and Jost L. Fischer (Inst. for Phys. and AstroPhys., Potsdam Univ., Karl/Liebknecht Str. 24-25, Potsdam 14469, Germany, markus.abel@physik.uni-potsdam.de)

Synchronization of musical instruments has raised attention due to the important implications on sound production in musical instruments and technological applications. In this contribution, we show new results on the interaction of two coupled organ pipes: we present a new experiment where the pipes were positioned in a plane with varying distance, further we briefly refer to a corresponding description in terms of a reduced model, and eventually show numerical simulations which are in full agreement with the measurements. Experimentally, the 2D setup allows for the observation of a new phenomenon: a synchronization/desynchronization transition at regular distances of the pipes. The developed model basically consists of a self-sustained oscillator with nonlinear, delayed coupling. The nonlinearity reflects the complicated interaction of emitted acoustical waves with the jet exiting at the organ pipe mouth, and the delay term takes care of the wave propagation. Synchronization is a clear evidence for the importance of nonlinearities in music and continues to be a source of astonishing results.

2:00

2pMU4. Nonlinear coupling mechanisms in acoustic oscillator systems which can lead to synchronization. Jost Fischer (Dept. for Phys. and Astronomy, Univ. of Potsdam, Karl-Liebknecht-Str 24/25, Potsdam, Brandenburg 14476, Germany, jost.fischer@uni-potsdam.de) and Markus Abel (Ambrosys GmbH, Potsdam, Germany)

We present results on the coupling mechanisms in wind-driven, self-sustained acoustic oscillators. Such systems are found in engineering applications, as gas burners, and—more beautiful—in musical instruments. As a result, we find that coupling and oscillators are nonlinear in character, which can lead to synchronization. We demonstrate our ideas using one of the oldest and most complex musical devices: organ pipes. Building up on the questions of preceding works, the elements of the sound generation are identified using detailed

experimental and theoretical studies, as well as numerical simulations. From these results, we derive the nonlinear coupling mechanisms of the mutual interaction of organ pipes. This leads to a nonlinear coupled acoustic oscillator model, which is based on the aeroacoustical and fluid dynamical first principles. The model calculations are compared with the experimental results from preceding works. It appears that the sound generation and the coupling mechanisms are properly described by the developed nonlinear coupled model of self-sustained oscillators. In particular, we can explain the unusual nonlinear shape Arnold tongues of the coupled two-pipe system. Finally, we show the power of modern CFD simulations by a 2D simulation of two mutually interacting organ pipes, i.e., the compressible Navier-Stokes equations are numerically solved.

2:20–2:40 Break

2:40

2pMU5. Auditory-inspired pitch extraction using a synchrony capture filterbank. Kumaresan Ramdas, Vijay Kumar Peddinti (Dept. of Elec., Comput. Eng., Univ. of Rhode Island, Kelley A216 4 East Alumni Ave., Kingston, RI 02881, kumar@ele.uri.edu), and Peter Cariani (Hearing Res. Ctr. & Dept. of Biomedical Eng., Boston Univ., Boston, MA)

The question of how harmonic sounds in speech and music produce strong, low pitches at their fundamental frequencies, F0's, has been of theoretical and practical interest to scientists and engineers for many decades. Currently the best auditory models for F0 pitch, (e.g., Meddis & Hewitt, 1991), are based on bandpass filtering (cochlear mechanics), half-wave rectification and low-pass filtering (hair cell transduction, synaptic transmission), channel autocorrelations (all-order interspike interval distributions) aggregated into a summary autocorrelation, followed by an analysis that determines the most prevalent interspike intervals. As a possible alternative to explicit autocorrelation computations, we propose an alternative model that uses an adaptive Synchrony Capture Filterbank (SCFB) in which channels in a filterbank neighborhood are driven exclusively (captured) by dominant frequency components closest to them. Channel outputs are then adaptively phase aligned with respect to a common time reference to compute a Summary Phase Aligned Function (SPAF), aggregated across all channels, from which F0 can then be easily extracted. Possible relations to brain rhythms and phase-locked loops are discussed. [Work supported by AFSOR FA9550-09-1-0119, Invited to special session about Synchronization in Musical Acoustics and Music Psychology.]

3:00

2pMU6. Identification of sound sources in soundscape using acoustic, psychoacoustic, and music parameters. Ming Yang and Jian Kang (School of Architecture, Univ. of Sheffield, Western Bank, Sheffield S10 2TN, United Kingdom, arp08my@sheffield.ac.uk)

This paper explores the possibility of automatic identification/classification of environmental sounds, by analyzing sound with a number of acoustic, psychoacoustic, and music parameters, including loudness, pitch, timbre, and rhythm. The sound recordings of single sound sources labeled in four categories, i.e., water, wind, birdsongs, and urban sounds including street music, mechanical sounds and traffic noise, are automatically identified with machine learning and mathematic models, including artificial neural networks and discriminant functions, based on the results of the psychoacoustic/music measures. The accuracies of the identification are above 90% for all the four categories. Moreover, based on these techniques, identification of construction noise sources from general urban background noise is explored, using the large construction project of London Bridge Station redevelopment as a case study site.

Contributed Papers

3:20

2pMU7. Neuronal synchronization of musical large-scale form. Lenz Hartmann (Insitut for Systematic Musicology, Universität Hamburg, Feldstrasse 59, Hamburg, Hamburg 20357, Germany, lenz.hartmann@gmx.de)

Musical form in this study is taken as structural aspects of music ranging over several bars as a combination of all elements that constitutes a piece of music, like pitch, rhythm or timbre. In an EEG-study 25 participants listen to the first about four minutes of a piece of electronic dance music for three times each and ERP grand-averages were calculated. Correlations of a one second time windows between the ERPs of all electrodes and therefore of different brain regions is used as a measure of synchronization between these areas. Local maxima corresponding to strong synchronization show up at expectancy points of boundaries in the present musical form. A modified FFT analysis of the ERPs of the mean of all trials and all channels that just take the ten highest peaks in consideration show strongest brain activity at frequencies in the gamma-band (about 40–60 Hz) and in the beta-band (about 20–30 Hz).

3:35

2pMU8. Nonlinearities and self-organization in the sound production of the Rhodes piano. Malte Muenster, Florian Pfeifle, Till Weinrich, and Martin Keil (Systematic Musicologie, Univ. of Hamburg, Pilatuspool, 19, Hamburg, Hamburg 20355, Germany, m.muenster@arcor.de)

Over the last five decades the Rhodes piano became a common keyboard instrument. It is played in such diverse musical genres as Jazz, Funk, Fusion, or Pop. The sound processing of the Rhodes has not been studied in detail beforeIts sound is produced by a mechanical driven tuning fork like system causing a change in the magnetic flux of an electromagnetic pick up system. The mechanical part of the tone production consists of a small diameter tine made of stiff spring steel, the tine, and a tone bar made of brass, which is strongly coupled to the former and acts as a resonator. The system is an example for strong generator-resonator coupling. The tine acts as a generator forcing the tonebar to vibrate with its fundamental frequency. Despite of extremely different and much lower eigenfrequencies the tonebar is enslaved by the tine. The tine is of lower spatial dimension and less damped and acts nearly linear. The geometry of the tonebar is much more complex and therefore of higher dimension and damped stronger. The vibration of these two parts are perfectly in-phase or anti-phase pointing to a quasi-synchronization behavior. Moreover, the tonebar is responsible for the timbre of the initial transient. It adds the glockenspiel sound to the transient and extends the sustain. The sound production is discussed as synergetic, self-organizing system, leading to a very precise harmonic overtone structure and characteristic initial transients enhancing the variety of musical performance.

Session 2pNSa

Noise and Psychological and Physiological Acoustics: New Frontiers in Hearing Protection II

Elliott H. Berger, Cochair
Occupational Health & Environmental Safety Division, 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650

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

Chair's Introduction—1:25

Invited Paper

1:30

2pNSa1. Comparison of impulse peak insertion loss measured with gunshot and shock tube noise sources. 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), Elliott H. Berger (Personal Safety Div., E-A-RCAL lab, 3M, Indianapolis, IN), and William A. Ahroon (US Army Aeromedical Res. Lab., US Army, Fort Rucker, AL)

The National Institute for Occupational Safety and Health in cooperation with scientists from 3M and the U.S. Army Aeromedical Research Laboratory conducted a series of Impulse peak insertion loss (IPIL) tests of the acoustic test fixtures from the Institute de Saint Louis (ISL) with a 0.223 caliber rifle and two different acoustic shock tubes. The Etymotic Research ETYPlugs™ earplug, 3M™ TacticalPro™ communication headset and the dual protector combination were tested with all three impulse noise sources. The spectra, IPIL, and the reduction of different damage risk criteria will be presented. The spectra from the noise sources vary considerably with the rifle having peak energy at about 1000 Hz. The shock tubes had peak levels around 125 and 250 Hz. The IPIL values for the rifle were greater than those measured with the two shock tubes. The shock tubes had comparable IPIL results except at 150 dB for the dual protector condition. The treatment of the double protection condition is complicated because the earmuff reduces the shock wave and reduces the effective level experienced by the earplug. For the double protection conditions, bone conduction presents a potential limiting factor for the effective attenuation that can be achieved by hearing protection.

Contributed Paper

1:50

2pNSa2. Evaluation of level-dependent performance of in-ear hearing protection devices using an enclosed sound source. Theodore F. Argo and G. Douglas Meegan (Appl. Res. Assoc., Inc., 7921 Shaffer Parkway, Littleton, CO 80127, targo@ara.com)

Hearing protection devices are increasingly designed with the capability to protect against impulsive sound. Current methods used to test protection from impulsive noise, such as blasts and gunshots, suffer from various drawbacks and complex, manual experimental procedures. For example, the use of a shock tube to emulate blast waves typically produces a blast wind of a much higher magnitude than that generated by an explosive, a specific but important inconsistency between the test conditions and final application. Shock tube test procedures are also very inflexible and provide only minimal insight into the function and performance of advanced electronic hearing protection devices that may have relatively complex response as a function of amplitude and frequency content. To address the issue of measuring the amplitude-dependent attenuation provided by a hearing protection device, a method using a compression driver attached to an enclosed waveguide was developed. The hearing protection device is placed at the end of the waveguide and the response to exposure to impulsive and frequency-dependent signals at calibrated levels is measured. Comparisons to shock tube and standard frequency response measurements will be discussed.

Invited Papers

2:05

2pNSa3. Exploration of flat hearing protector attenuation and sound detection in noise. Christian Giguere (Audiology/SLP Program, Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H8M5, Canada, cgiguere@uottawa.ca) and Elliott H. Berger (Personal Safety Div., 3M, Indianapolis, IN)

Flat-response devices are a class of hearing protectors with nearly uniform attenuation across frequency. These devices can protect the individual wearer while maintaining the spectral balance of the surrounding sounds. This is typically achieved by reducing the muffling effect of conventional hearing protectors which provide larger attenuation at higher than lower frequencies, especially with earmuffs. Flat hearing protectors are often recommended when good speech communication or sound perception is essential, especially for wearers with high-frequency hearing loss, to maintain audibility at all frequencies. However, while flat-response devices are described in some acoustical standards, the tolerance limits for the definition of flatness are largely unspecified and relatively little is known on the exact conditions when such devices can be beneficial. The purpose of this study is to gain insight into the interaction between the spectrum of the noise, the shape of the attenuation-frequency response, and the hearing loss configuration on detection thresholds using a psychoacoustic model of sound detection in noise.

2:25

2pNSa4. Electronic sound transmission hearing protectors and horizontal localization: Training/adaptation effects. John Casali (Auditory Systems Lab, Virginia Tech, 250 Durham Hall, Blacksburg, VA 24061, jcasali@vt.edu) and Martin Robinette (U.S. Army Public Health Command, U.S. Army, Aberdeen Proving Ground, MD)

Auditory situation awareness is known to be affected by some hearing protectors, even advanced electronic devices. A horizontal localization task was employed to determine how use/training with electronic sound transmission hearing protectors affected auditory localization ability, as compared to open-ear. Twelve normal-hearing participants performed baseline localization testing in a hemi-anechoic field in three listening conditions: open-ear, in-the-ear (ITE) device (Etymotic EB-15), and over-the-ear (OTE) device (Peltor ComTac II). Participants then wore either the ITE or OTE protector for 12, almost daily, one-hour training sessions. Post-training, participants again underwent localization testing with all three conditions. A computerized custom software-hardware interface presented localization sounds and collected accuracy and timing measures. ANOVA and post hoc statistical tests revealed that pre-training localization performance with either the ITE or OTE protector was significantly worse ($p<0.05$) than open-ear performance. After training with any given listening condition, performance in that condition improved, in part from a practice effect. However, post-training localization showed near equal performance between the open-ear and the protector on which training occurred. Auditory learning, manifested as significant localization accuracy improvement, occurred for the training device, but not for the non-training device, i.e., no crossover benefit from the training device to the non-training device occurred.

Contributed Papers

2:45

2pNSa5. Measuring effective detection and localization performance of hearing protection devices. Richard L. McKinley (Battlespace Acoust., Air Force Res. Lab., 2610 Seventh St., AFRL/711HPW/RHCB, Wright-Patterson AFB, OH 45433-7901, richard.mckinley.1@us.af.mil), Eric R. Thompson (Ball Aerosp. and Technologies, Air Force Res. Lab., Wright-Patterson AFB, OH), and Brian D. Simpson (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH)

Awareness of the surrounding acoustic environment is essential to the safety of persons. However, the use of hearing protection devices can degrade the ability to detect and localize sounds, particularly quiet sounds. There are ANSI/ASA standards describing methods for measuring attenuation, insertion loss, and speech intelligibility in noise for hearing protection devices, but currently there are no standard methods to measure the effects of hearing protection devices on localization and/or detection performance. A method for measuring the impact of hearing protectors on effective detection and localization performance has been developed at AFRL. This method measures the response time in an aurally aided visual search task where the presentation levels are varied. The performance with several level-dependent hearing protection devices will be presented.

3:00

2pNSa6. Personal alert safety system localization field tests with firefighters. Joelle I. Suits, Casey M. Farmer, Ofodike A. Ezekoye (Dept. of Mech. Eng., The Univ. of Texas at Texas, 204 E Dean Keaton St., Austin, TX 78712, jsuits@utexas.edu), Mustafa Z. Abbasi, and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

When firefighters get lost or incapacitated on the fireground, there is little time to find them. This project has focused on a contemporary device used in this situation, the Personal Alert Safety System. We have studied the noises on the fireground (i.e., chainsaws, gas powered ventilation fans, pumper trucks) [J. Acoust. Soc. Am. **134**, 4221 (2013)], how the fire environment affects sound propagation [J. Acoust. Soc. Am. **134**, 4218 (2013)], and how firefighter personal protective equipment (PPE) affects human hearing [POMA **19**, 030054 (2013)]. To put all these pieces together, we have traveled to several fire departments across the country conducting tests to investigate how certain effects manifest themselves when firefighters search for the source of a sound. We tasked firefighters to locate a target sound in various acoustic environments while their vision was obstructed and while wearing firefighting PPE. We recorded how long it took them to find the source, what path they took, when they first heard the target sound, and the frequency content and sound pressure level of the acoustic environment. The results will be presented in this talk. [Work supported by U.S. Department of Homeland Security Assistance to Firefighters Grants Program.]

3:15

2pNSa7. Noise level from burning articles on the fireground. Mustafa Z. Abbasi, Preston S. Wilson (Appl. Res Lab and Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78751, mustafa_abbasi@utexas.edu), and Ofodike A. Ezekoye (Dept. of Mech. of Eng., The Univ. of Texas at Austin, Austin, TX)

Firefighters encounter an extremely difficult environment due to the presence of heat, smoke, falling debris etc. If one of them needs rescue, an audible alarm is used to alert others of their location. This alarm, known as the Personal Alert Safety System (PASS) alarm, has been part of firefighter gear since the early 1980s. The PASS has been enormously successful, but a review of The National Institute for Occupational Safety and Health (NIOSH) firefighter fatality report suggests that there are instances when the alarm is not heard or not localized. In the past, we have studied fireground noise from various pieces of gear such as chainsaws and fans, etc., to understand the soundscape present during a firefighting operation. However, firefighters, and other interested parties have raised the issue of noise caused by the fire itself. The literature shows that buoyancy-controlled, non-premixed flames aerodynamically oscillate in the 10–16 Hz range, depending on the diameter of the fuel base. Surprisingly, few acoustic measurements have been made even for these relatively clean fire conditions. However, experience suggests burning items do create sounds. Most likely these sound are from the decomposition of the material as it undergoes pyrolysis (turns in gaseous fuel and char). This paper will present noise measurements from various burning articles as well as characterization of the fire to understand this noise source.

3:30

2pNSa8. Bacterial attachment and insertion loss of earplugs used long-time in the noisy workplace. Jinro Inoue, Aya Nakamura, Yumi Tanizawa, and Seichi Horie (Dept. of Health Policy and Management, Univ. of Occupational and Environ. Health, Japan, 1-1 Iseigaoka, Yahatanishi-ku, Kitakyushu, Fukuoka 807-8555, Japan, j-inoue@med.uoeh-u.ac.jp)

In the real noisy workplace, workers often use earplugs for a longtime. We assessed the condition of bacterial attachment and the insertion loss of 197 pairs of earplugs collected from 6 companies. The total viable counts and the presence of *Staphylococcus aureus* were examined by 3M Petrifilm. The insertion losses were evaluated by GRAS 45CB Acoustic Test Fixture. We detected greater number of viable counts in the foam earplugs than in the premolded earplugs. Staphylococcus aureus was detected in 10 foam earplugs (5.1%). The deterioration of insertion loss was found only in the deformed earplugs. The condition of work environment such as presence of dust or use of oily liquid might cause the deterioration. Both the condition of bacterial attachment and the insertion loss were not correlated with the duration of use. We observed no correlation between the condition of bacterial attachment and the insertion loss of earplugs and neither of them was related to the duration of long-term use of the earplugs.

TUESDAY AFTERNOON, 28 OCTOBER 2014 MARRIOTT 9/10, 1:00 P.M. TO 4:20 P.M.

Session 2pNSb

Noise and Structural Acoustics and Vibration: Launch Vehicle Acoustics II

R. Jeremy Kenny, Cochair
Marshall Flight Center, NASA, Huntsville, AL 35812

Tracianne B. Neilsen, Cochair
Brigham Young University, N311 ESC, Provo, UT 84602

Chair's Introduction—1:00

Invited Papers

1:05

2pNSb1. Comparison of the acoustical emissions of multiple full-scale rocket motors. Michael M. James, Alexandria R. Salton (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, michael.james@blueridgeresearch.com), Kent L. Gee, and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Development of the next-generation space flight vehicles has prompted a renewed focus on rocket sound source characterization and near-field propagation modeling. Improved measurements of the sound near the rocket plume are critical for direct determination of the acoustical environment both in the near and far-fields. They are also crucial inputs to empirical models and to validate computational aeroacoustics models. Preliminary results from multiple measurements of static horizontal firings of Alliant Techsystems motors including the GEM-60, Orion 50S XLG, and the Reusable Solid Rocket Motor (RSRM) performed in Promontory, UT, are analyzed and compared. The usefulness of scaling by physical parameters such as nozzle diameter, velocity, and overall sound power is demonstrated. The sound power spectra, directional characteristics, distribution along the exhaust flow, and pressure statistical metrics are examined over the multiple motors. These data sets play an important role in formulating more realistic sound source models, improving acoustic load estimations, and aiding in the development of the next generation space flight vehicles via improved measurements of sound near the rocket plume.

1:25

2pNSb2. Low-dimensional acoustic structures in the near-field of clustered rocket nozzles. Andres Canchero, Charles E. Tinney (Aerosp. Eng. and Eng. Mech., The Univ. of Texas at Austin, 210 East 24th St., WRW-307, 1 University Station, C0600, Austin, TX 78712-0235, andres.canchero@utexas.edu), Nathan E. Murray (National Ctr. for Physical Acoust., Univ. of MS, Oxford, MS), and Joseph H. Ruf (NASA Marshall Space Flight Ctr., Huntsville, AL)

The plume and acoustic field produced by a cluster of two and four rocket nozzles is visualized by way of retroreflective shadowgraphy. Both steady state and transient operations of the nozzles (start-up and shut-down) were conducted in the fully-anechoic chamber and open jet facility of The University of Texas at Austin. The laboratory scale rocket nozzles comprise thrust-optimized parabolic (TOP) contours, which during start-up, experience free shock separated flow, restricted shock separated flow, and an "end-effects regime" prior to flowing full. Shadowgraphy images are first compared with several RANS simulations during steady operations. A proper orthogonal decomposition (POD) of various regions in the shadowgraphy images is then performed to elucidate the prominent features residing in the supersonic annular flow region, the acoustic near field and the interaction zone that resides between the nozzle

plumes. Synchronized surveys of the acoustic loads produced in close vicinity to the rocket clusters are compared to the low-order shadowgraphy images in order to identify the various mechanisms within the near-field that are responsible for generating sound.

1:45

2pNSb3. Experimental study on lift-off acoustic environments of launch vehicles by scaled cold jet. Hiroki Ashida, Yousuke Takeyama (Integrated Defence & Space Systems, Mitsubishi Heavy Industries, Ltd., 10, Oye-cho, Minato-ku, Nagoya City, Aichi 455-8515, Japan, hiroki1_ashida@mhi.co.jp), Kiyotaka Fujita, and Aki Azusawa (Technol. & Innovation Headquarters, Mitsubishi Heavy Industries, Ltd., Aichi, Japan)

Mitsubishi Heavy Industries (MHI) have been operating the current Japanese flagship launch vehicle H-IIA and H-IIB, and developing the next flagship launch vehicle H-X. The concept of H-X is affordable, convenient, and comfortable for payloads including mitigation of acoustic environment during launch. Acoustic measurements were conducted using scaled GN2 cold jet and aperture plate to facilitate understanding of lift-off acoustic source and to take appropriate measures against it without use of water injection. It was seen that the level of vehicle acoustics in high frequency range depends on the amount of interference between the jet and the plate, and enlargement of the aperture is effective for acoustic mitigation.

2:05

2pNSb4. Detached-Eddy simulations of rocket plume noise at lift-off. A. Lyrintzis, V. Golubev (Aerosp. Eng., Embry-Riddle Aeronutical Univ., Daytona Beach, FL), K. Kurbatski (Aerosp. Eng., Embry-Riddle Aeronutical Univ., Lebanon, New Hampshire), E. Osman (Aerosp. Eng., Embry-Riddle Aeronutical Univ., Denver, Colorado), and Reda Mankbadi (Aerosp. Eng., Embry-Riddle Aeronutical Univ., 600 S. Clyde Morris Blvd, Daytona Beach, FL 32129, Reda.Mankbadi@erau.edu)

The three-dimensional turbulent flow and acoustic field of a supersonic jet impinging on a solid plate at different inclination angles is studied computationally using the general-purpose CFD code ANSYS Fluent. A pressure-based coupled solver formulation with the second-order weighted central-upwind spatial discretization is applied. Hot jet thermal condition is considered. Acoustic radiation of impingement tones is simulated using a transient time-domain formulation. The effects of turbulence in steady state are modeled by the SST k- turbulence model. The Wall-Modeled Large-Eddy Simulation (WMLES) model is applied to compute transient solutions. The near-wall mesh on the impingement plate is fine enough to resolve the viscosity-affected near-wall region all the way to the laminar sublayer. Inclination angle of the impingement plate is parameterized in the model for automatic re-generation of the mesh and results. The transient solution reproduces the mechanism of impingement tone generation by the interaction of large-scale vortical structures with the impingement plate. The acoustic near field is directly resolved by the Computational Aeroacoustics (CAA) to accurately propagate impingement tone waves to near-field microphone locations. Results show the effect of the inclination angle on sound level pressure spectra and overall sound pressure level directivities.

2:25

2pNSb5. Large-eddy simulations of impinging over-expanded supersonic jet noise for launcher applications. Julien Troyes, François Vuillot (DSNA, Onera, BP72, 29 Ave. de la Div. Leclerc, Châtillon Cedex 92322, France, julien.troyes@onera.fr), and Hadrien Lambaré (DLA, CNES, Paris, France)

During the lift-off phase of a space launcher, powerful rocket motors generate harsh acoustic environment on the launch pad. Following the blast waves created at ignition, jet noise is a major contributor to the acoustic loads received by the launcher and its payload. Recent simulations performed at ONERA to compute the noise emitted by solid rocket motors at lift-off conditions are described. Far-field noise prediction is achieved by associating a LES solution of the jet flow with an acoustics surface integral method. The computations are carried out with in-house codes CEDRE for the LES solution and KIM for Ffowcs Williams & Hawkings porous surface integration method. The test case is that of a gas generator, fired vertically onto a 45 degrees inclined flat plate which impingement point is located 10 diameters from nozzle exit. Computations are run for varied numerical conditions, such as turbulence modeling along the plate and different porous surfaces location and type. Results are discussed and compared with experimental acoustic measurements obtained by CNES at MARTEL facility.

2:45–3:05 Break

3:05

2pNSb6. Scaling metrics for predicting rocket noise. Gregory Mack, Charles E. Tinney (Ctr. for AeroMech. Res., The Univ. of Texas at Austin, ASE/EM, 210 East 24th St., Austin, TX 78712, cetinney@utexas.edu), and Joseph Ruf (Combustion and Flow Anal. Team, ER42, NASA Marshal Space Flight Ctr., Huntsville, AL)

Several years of research at The University of Texas at Austin concerning the sound field produced by large area-ratio rocket nozzles is presented [Baars *et al*., AIAA J. 50(1), (2012); Baars and Tinney, Exp. Fluids, 54 (1468), (2013); Donald *et al*., AIAA J. 52(7), (2013)]. The focus of these studies is on developing an in-depth understanding of the various acoustic mechanisms that form during start-up of rocket engines and how they may be rendered less efficient in the generation of sound. The test articles comprise geometrically scaled replicas of large area ratio nozzles and are tested in a fully anechoic chamber under various operating conditions. A framework for scaling laboratory-scale nozzles is presented by combining established methods with new methodologies [Mayes, NASA TN D-21 (1959); Gust, NASA TN-D-1999 (1964); Eldred, NASA SP-8072 (1972); Sutherland AIAA Paper 1993–4383 (1993); Varnier, AIAA J. 39:10 (2001); James *et al*. Proc. Acoust. Soc. Amer. 18(3aNS), (2012)]. In particular, both hot and cold flow tests are reported which comprise single, three and four nozzle clusters. An effort to correct for geometric scaling is also presented.

3:25

2pNSb7. Acoustic signature characterization of a sub-scale rocket launch. David Alvord and K. Ahuja (Aerosp. & Acoust. Technologies Div., Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, david.alvord@gtri.gatech.edu)

Georgia Tech Research Institute (GTRI) conducted a flight test of a sub-scale rocket in 2013 outside Talladega, Alabama, to acquire the launch acoustics produced. The primary objective of the test was to characterize the acquired data during a sub-scale launch and compare it with heritage launch data from the STS-1 Space Shuttle flight. Neither launch included acoustic suppression; however, there were differences in the ground geometry. STS-1 launched from the Mobile Launch Platform at Pad 39B with the RS-25 liquid engines and Solid Rocket Boosters (SRBs) firing into their respective exhaust ducts and flame trench, while the GTRI flight test vehicle launched from a flat reflective surface. The GTRI launch vehicle used a properly scaled Solid Rocket Motor (SRM) for propellant; therefore, primary analysis will focus on SRM/SRB centric acoustic events. Differences in the Ignition Overpressure (IOP) wave signature between both due to this will be addressed. Additionally, the classic liftoff acoustics "football shape" is preserved between both full and sub-scale flights. The launch signatures will be compared, with note taken of specific launch acoustic events more easily investigated with sub-scale launch data or supplement current sub-scale static hotfire testing.

3:45

2pNSb8. Infrasonic energy from orbital launch vehicles. W. C. Kirkpatrick Alberts, John M. Noble, and Stephen M. Tenney (US Army Res.Lab., 2800 Powder Mill, Adelphi, MD 20783, kirkalberts@verizon.net)

Large, heavy-lift rockets have significant acoustic and infrasonic energy that can often be detected from a considerable distance. These sounds, under certain environmental conditions, can propagate hundreds of kilometers from the launch location. Thus, ground-based infrasound arrays can be used to monitor the low frequencies emitted by these large rocket launches. Multiple launches and static engine tests have been successfully recorded over many years using small infrasound arrays at various distances from the launch location. Infrasonic measurements using a 20 m array and parabolic equation modeling of a recent launch of an Aries V rocket at Wallops Island, Virginia, will be discussed.

Contributed Paper

4:05

2pNSb9. Influence of source level, peak frequency, and atmospheric absorption on nonlinear propagation of rocket noise. Michael F. Pearson (Phys., Brigham Young Univ., 560 W 700 S, Lehi, UT 84043, m3po22@gmail.com), Kent L. Gee, Tracianne B. Neilsen, Brent O. Reichman (Phys., Brigham Young Univ., Provo, UT), Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC), and Alexandira R. Salton (Blue Ridge Res. and Consulting, Asheville, Utah)

Nonlinear propagation effects in rocket noise have been previously shown to be significant [M. B. Muhlestein *et al.* Proc. Mtgs. Acoust. (2013)]. This paper explores the influence of source level, peak frequency, and ambient atmospheric conditions on predictions of nonlinear propagation. An acoustic pressure waveform measured during a full-scale solid rocket motor firing is numerically propagated via generalized Burgers equation model for atmospheric conditions representative of plausible spaceport locations. Cases are explored where the overall sound pressure level and peak frequency has been scaled to model engines of different scale or thrust. The predicted power spectral densities and overall sound pressure levels, both flat and A-weighted, are compared for nonlinear and linear propagation for distances up to 30 km. The differences in overall level suggest that further research to appropriately include nonlinear effects in launch vehicle noise models is worthwhile.

Session 2pPA

Physical Acoustics and Education in Acoustics: Demonstrations in Acoustics

Uwe J. Hansen, Cochair
Chemistry & Physics, Indiana State University, 64 Heritage Dr., Terre Haute, IN 47803-2374

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

Chair's Introduction—1:00

Invited Papers

1:05

2pPA1. Sharing experiments for home and classroom demonstrations. Thomas D. Rossing (Stanford Univ., Music, Stanford, CA 94305, rossing@ccrma.stanford.edu)

In the third edition of The Science of Sound, we included a list of "Experiments for Home, Laboratory, and Classroom Demonstrations" at the end of each chapter. Some of the demonstrations are done by the instructor in class, some are done by students for extra credit, some are intended to be done at home. We describe a representative number of these, many of which can be done without special equipment.

1:30

2pPA2. A qualitative demonstration of the behavior of the human cochlea. Andrew C. Morrison (Dept. of Natural Sci., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

Demonstrations of the motion of the basilar membrane in the human cochlea designed by Keolian [J. Acoust. Soc. Am. **101**, 1199-1201 (1997)], Tomlinson et. al. [J. Acoust. Soc. Am. **121**, 3115 (2007)], and others provide a way for students in a class to visualize the behavior of the basilar membrane and explore the physical mechanisms leading to many auditory phenomena. The designs of Keolian and Tomlinson are hydrodynamic. A non-hydrodynamic apparatus has been designed that can be constructed with commonly available laboratory supplies and items readily available at local hardware stores. The apparatus is easily set up for demonstration purposes and is compact for storing between uses. The merits and limitations of this design will be presented.

1:55

2pPA3. Nonlinear demonstrations in acoustics. Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

The world is nonlinear, and in presenting demonstrations in acoustics, one often has to consider the effects of nonlinearity. In this presentation the nonlinear effects are made to be very pronounced. The nonlinear effects of standing waves on a lightly stretched string (which is also very elastic) lead to wave shape distortion, mode jumping and hysteresis effects in the resonant behavior of a tuning curve near a resonance. The effects of hyperelasticity in a rubber string are discussed. A two dimensional system like a vibrating rectangular or circular drum-head are well known. The nonlinear effects of standing waves on a lightly stretched hyperelastic membrane make an interesting and challenging study. Here, tuning curve behavior demonstrates that there is softening of the system for slightly increasing vibration amplitudes followed by stiffening of the system at larger vibration amplitudes. The hysteretic behavior of the tuning curve for sweeping from lower to higher frequencies and then from higher to lower frequencies (for the same drive amplitude) is demonstrated. Lastly, the nonlinear effects of a column of soil or fine granular material loading a thin elastic circular clamped plate are demonstrated near resonance. Here again, the nonlinear highly asymmetric tuning curve behavior is demonstrated.

2:20–2:40 Audience Interaction

Session 2pSA

Structural Acoustics and Vibration, Signal Processing in Acoustics, and Engineering Acoustics: Nearfield Acoustical Holography

Sean F. Wu, Chair
Mechanical Engineering, Wayne State University, 5050 Anthony Wayne Drive, College of Engineering Building, Rm 2133, Detroit, MI 48202

Chair's Introduction—2:00

Invited Papers

2:05

2pSA1. Transient nearfield acoustical holography. Sean F. Wu (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., College of Eng. Bldg., Rm. 2133, Detroit, MI 48202, sean_wu@wayne.edu)

Transient Nearfield Acoustical Holography Sean F. Wu Department of Mechanical Engineering, Wayne State University, Detroit, MI, 48202 This paper presents the general formulations for reconstructing the transient acoustic field generated by an arbitrary object with a uniformly distributed surface velocity in free space. These formulations are derived from the Kirchhoff-Helmholtz integral theory that correlates the transient acoustic pressure at any field point to those on the source surface. For a class of acoustic radiation problems involving an arbitrarily oscillating object with a uniformly distributed surface velocity, for example, a loudspeaker membrane, the normal surface velocity is frequency dependent but is spatially invariant. Accordingly, the surface acoustic pressure is expressible as the product of the surface velocity and the quantity that can be solved explicitly by using the Kirchhoff-Helmholtz integral equation. This surface acoustic pressure can be correlated to the field acoustic pressure using the Kirchhoff-Helmholtz integral formulation. Consequently, it is possible to use nearfield acoustic holography to reconstruct acoustic quantities in entire three-dimensional space based on a single set of acoustic pressure measurements taken in the near field of the target object. Examples of applying these formulations to reconstructing the transient acoustic pressure fields produced by various arbitrary objects are demonstrated.

2:30

2pSA2. A multisource-type representation statistically optimized near-field acoustical holography method. Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

A reduced-order approach to near-field acoustical holography (NAH) that accounts for sound fields generated by multiple spatially separated sources of different types is presented. In this method, an equivalent wave model (EWM) of a given field is formulated based on rudimentary knowledge of source types and locations. The statistically optimized near-field acoustical holography (SONAH) algorithm is utilized to perform the NAH projection after the formulation of the multisource EWM. The combined process is called multisource-type representation SONAH (MSTR SONAH). This method is used to reconstruct simulated sound fields generated by combinations of multiple source types. It is shown that MSTR SONAH can successfully reconstruct the near field pressures in multisource environments where other NAH methods result in large errors. The MSTR SONAH technique can be extended to general sound fields where the shapes and locations of sources and scattering bodies are known.

2:55

2pSA3. Bayesian regularization applied to real-time near-field acoustic holography. Thibaut Le Magueresse (MicrodB, 28 chemin du petit bois, Ecully 69131, France, thibaut-le-magueresse@microdb.fr), Jean-Hugh Thomas (Laboratoire d'Acoustique de l'Université du Maine, Le Mans, France), Jérôme Antoni (Laboratoire Vibrations Acoustique, Villeurbanne, France), and Sébasien Paillasseur (MicrodB, Ecully, France)

Real-Time Near-field Acoustic Holography is used to recover non stationary acoustic sound sources using a planar microphone array. In the direct way, describing propagation requires the convolution of the spatial spectrum of the source under study with a known impulse response. When the convolution operator is replaced with a matrix product, the propagation operator is re-written in a Toeplitz matrix form. Solving the inverse problem is based on a Singular value decomposition of this propagator and Tikhonov regularization is used to stabilize the solution. The purpose here is to study the regularization process. The formulation of this problem in the Tikhonov sense estimates the solution from the knowledge of the propagation model, the measurements and the regularization parameter. This parameter is calculated by making a compromise between the fidelity to the real measured data and the fidelity to available a priori information. A new regularization parameter is introduced based on a Bayesian approach to maximize the information taken into account. Comparisons of the results are proposed, using the L-Curve and the generalized cross validation. The superiority of the Bayesian parameter is observed for the reconstruction of a non stationary experimental source using real-time near-field acoustic holography.

Contributed Papers

3:20

2pSA4. Acoustic building infiltration measurement system. Ralph T. Muehleisen, Eric Tatara (Decision and Information Sci., Argonne National Lab., 9700 S. Cass Ave., Bldg. 221, Lemont, IL 60439, rmuehleisen@anl.gov), Ganesh Raman, and Kanthasamy Chelliah (Mater., Mech., and Aerosp. Eng., Illinois Inst. of Technol., Chicago, IL)

Building infiltration is a significant portion of the heating and cooling load of buildings and accounts for nearly 4% of the total energy use in the United States. Current measurement methods for locating and quantifying infiltration in commercial buildings to apply remediation are very limited. In this talk, the development of a new measurement system, the Acoustic Building Infiltration Measurement System (ABIMS) is presented. ABIMS uses Nearfield Acoustic Holography (NAH) to measure the sound field transmitted through a section of the building envelope. These data are used to locate and quantify the infiltration sites of a building envelope section. The basic theory of ABIMS operation and results from computer simulations are presented.

3:35

2pSA5. Reversible quasi-holographic line-scan processing for acoustic imaging and feature isolation of transient scattering. Daniel Plotnick, Philip L. Marston, David J. Zartman (Phys. and Astronomy, Washington State Univ., 1510 NW Turner DR, Apartment 4, Pullman, WA 99163, dsplotnick@gmail.com), and Timothy M. Marston (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Transient acoustic scattering data from objects obtained using a one-dimensional line scan or two-dimensional raster scan can be processed via a linear quasi-holographic method [K. Baik, C. Dudley, and P. L. Marston, J. Acoust. Soc. Am. 130, 3838–3851 (2011)] in a way that is reversible, allowing isolation of spatially or temporally dependent features [T. M. Marston *et al.*, in Proc. IEEE Oceans 2010]. Unlike nearfield holography the subsonic wavenumber components are suppressed in the processing. Backscattering data collected from a collocated source/receiver (monostatic scattering) and scattering involving a stationary source and mobile receiver (bistatic) may be processed in this manner. Distinct image features such as those due to edge diffraction, specular reflection, and elastic effects may be extracted in the image domain and then reverse processed to allow examination of those features in time and spectral domains. Multiple objects may also be isolated in this manner and clutter may be removed [D. J. Zartman, D. S. Plotnick, T. M. Marston, and P. L. Marston, Proceedings of Meetings on Acoustics 19, 055011 (2013) http://dx.doi.org/10.1121/1.4800881]. Experimental examples comparing extracted features with physical models will be discussed and demonstrations of signal enhancement in an at sea experiment, TREX13, will be shown. [Work supported by ONR.]

TUESDAY AFTERNOON, 28 OCTOBER 2014 — MARRIOTT 5, 1:00 P.M. TO 5:00 P.M.

Session 2pSC

Speech Communication: Segments and Suprasegmentals (Poster Session)

Olga Dmitrieva, Chair
Purdue University, 640 Oval Drive, Stanley Coulter 166, West Lafayette, IN 47907

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

Contributed Papers

2pSC1. Interactions among lexical and discourse characteristics in vowel production. Rachel S. Burdin, Rory Turnbull, and Cynthia G. Clopper (Linguist, The Ohio State Univ., 1712 Neil Ave., 222 Oxely Hall, Columbus, OH 43210, burdin@ling.osu.edu)

Various factors are known to affect vowel production, including word frequency, neighborhood density, contextual predictability, mention in the discourse, and audience. This study explores interactions between all five of these factors on vowel duration and dispersion. Participants read paragraphs that contained target words which varied in predictability, frequency, and density. Each target word appeared twice in the paragraph. Participants read each paragraph twice: as if they were talking to a friend ("plain speech") and as if they were talking to a hearing-impaired or non-native interlocutor ("clear speech"). Measures of vowel duration and dispersion were obtained. Results from the plain speech passages revealed that second mention and more predictable words were shorter than first mention and less predictable words, and that vowels in second mention and low density words were less peripheral than in first mention and high density words. Interactions between frequency and mention, and density and mention, were also observed, with second mention reduction only occurring in low density and low frequency words. We expect to observe additional effects of speech style, with clear speech vowels being longer and more disperse than plain speech vowels, and that these effects will interact with frequency, density, predictability, and mention.

2pSC2. Phonetic correlates of phonological quantity of Yakut. Lena Vasilyeva, Juhani Järvikivi, and Anja Arnhold (Dept. of Linguist, Univ. of AB, Edmonton, AB T6G2E7, Canada, lvasilye@ualberta.ca)

We investigated vowel quantity in Yakut (Sakha), a Turkic language spoken in Siberia by over 400,000 speakers in the Republic of Sakha (Yakutia) in the Russian Federation. Yakut is a quantity language; all vowel and consonant phonemes have short and long contrastive counterparts. The study aims at revealing acoustic characteristics of the binary quantity distinction in vowels. We used two sets of data: (1) A female native Yakut speaker read a 200-word list containing disyllabic nouns and verbs with four different combinations of vowel length in the two syllables (short–short, short–long, long–short, and long–long) and a list of 50 minimal pairs differing only in vowel length; (2) Spontaneous speech data from 9 female native Yakut speakers (aged 19–77), 200 words with short vowels and 200 words with long vowels, were extracted for analysis. Acoustic measurements of the short and long vowels' f0-values, duration and intensity were done. Mixed-effects models showed a significant durational difference between long and short vowels for both data sets. However, the preliminary results indicated that, unlike in quantity languages like Finnish and Estonian, there was no consistent effect of f0 as the phonetic correlate in Yakut vowel quantity distinction.

2pSC3. Acoustic and perceptual characteristics of vowels produced by self-identified gay and heterosexual male speakers. Keith Johnson (Linguist, Univ. of California, Berkeley, Berkeley, CA) and Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Prior research (Tracy & Satariano, 2011) investigated the perceptual characteristics of gay and heterosexual male speech; it was discovered that listeners primarily relied on vowels to identify sexual orientation. Using single-word utterances produced by those same speakers, the current study examined both the acoustic characteristics of vowels, such as pitch, duration, and the size of the vowel space, and how these characteristics relate to the perceived sexual orientation of the speaker. We found a correlation between pitch and perceived sexual identity for vowels produced by heterosexual speakers—higher f0 was associated with perceptual "gayness." We did not find this correlation for gay speakers. Vowel duration did not reliably distinguish gay and heterosexual speakers, but speakers who produced longer vowels were perceived as gay and speakers who produced shorter vowels were perceived as heterosexual. The size of the vowel space did not reliably differ between gay and heterosexual speakers. However, speakers who produced a larger vowel space were perceived as more gay-sounding than speakers who produced a smaller vowel space. The results suggest that listeners rely on these acoustic characteristics when asked to determine a male speaker's sexual orientation, but that the stereotypes that they seem to rely upon are inaccurate.

2pSC4. Acoustic properties of the vowel systems of Bolivian Quechua/Spanish bilinguals. Nicole Holliday (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, nrh245@nyu.edu)

This paper describes the vowel systems of Quechua/Spanish bilinguals in Cochabamba, Bolivia, and examines these systems to illuminate variation between phonemic and allophonic vowels in this Quechua variety. South Bolivian Quechua is described as phonemically trivocalic, and Bolivian Spanish is described as pentavocalic (Cerrón-Palomino 1994). Although South Bolivian Quechua has three vowel categories, Quechua uvular stop consonants promote high vowel lowering, effectively lowering /i/ and /u/ towards space otherwise occupied by /e/ and /o/ respectively, producing a system with five surface vowels but three phonemic vowels (Buckley 2000). The project was conducted with eleven Quechua/Spanish bilinguals from the Cochabamba department in Bolivia. Subjects participated in a Spanish to Quechua oral translation task and a word list task in Spanish. Results indicate that Quechua/Spanish bilinguals maintain separate vowel systems. In the Spanish vowel systems, each vowel occupies its own space and backness. A one-way ANOVA reveals that /i/ is higher and fronter than /e/, and /u/ is higher than /o/ ($p<0.05$). The Quechua vowel systems are somewhat more variable, with substantial overlap between /i/ and /e/, and between /u/ and /o/. Potential explanations for this result include lexical conditioning, speaker literacy effects, and differences in realizations of phonemic versus allophonic vowels.

2pSC5. Cue integration in the perception of fricative-vowel coarticulation in Korean. Goun Lee and Allard Jongman (Linguist, The Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045-3129, cconni@ku.edu)

Korean distinguishes two fricatives—fortis [s'] and non-fortis [s]. Perception of this distinction was tested in two different vowel contexts, with three types of stimuli (consonant-only, vowel-only, or consonant-vowel sequences) (Experiment 1). The relative contribution of consonantal and vocalic cues was also examined with cross-spliced stimuli (Experiment 2). Listeners' weighting of 7 perceptual cues—spectral mean (initial 60%, final 40%), vowel duration, H1-H2* (onset, mid), and cepstral peak prominence (onset, mid)—was examined. The data demonstrate that identification performance was heavily influenced by vowel context and listener performance was more accurate in the /a/ vowel context than in the /i/ vowel context. In addition, the type of stimulus presented changed the perceptual cue weighting. When presented with conflicting cues, listener decisions were driven by the vocalic cues in the /a/ vowel context. These results suggest that perceptual cues associated with breathy phonation are the primary cues for fricative identification in Korean.

2pSC6. Voicing, devoicing, and noise measures in Shanghainese voiced and voiceless glottal fricatives. Laura L. Koenig (Haskins Labs and Long Island Univ., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu) and Lu-Feng Shi (Haskins Labs and Long Island Univ., Brooklyn, New York)

Shanghainese has a rather rare voicing distinction between the glottal fricatives /h/ and /ɦ/. We evaluate the acoustic characteristics of this contrast in ten male and ten female speakers of urban Shanghainese dialect. Participants produced 20 CV words with a mid/low central vowel in a short carrier phrase. All legal consonant-tone combinations were used: /h/ preceded high, low, and short tones whereas /ɦ/ preceded low and short tones. Preliminary analyses suggested that the traditional "voiced" and "voiceless" labels for these sounds are not always phonetically accurate; hence we measure the duration of any voicing break relative to the entire phrase, as well as the harmonics-to-noise ratio (HNR) over the time. We expect longer relative voiceless durations and lower HNR measures for /h/ compared to /ɦ/. A question of interest is whether any gender differences emerge. A previous study on American English [Koenig, 2000, JSLHR 43, 1211–1228] found that men phonated through their productions of /h/ more often than women, and interpreted that finding in terms of male-female differences in vocal fold characteristics. A language that contrasts /h/ and /ɦ/ might minimize any such gender variation. Alternatively, the contrast might be realized in slightly different ways in men and women.

2pSC7. Incomplete neutralization of sibilant consonants in Penang Mandarin: A palatographic case study. Ting Huang, Yueh-chin Chang, and Feng-fan Hsieh (Graduate Inst. of Linguist, National Tsing Hua Univ., Rm. B306, HSS Bldg., No. 101, Section 2, Kuang-Fu Rd., Hsinchu City 30013, Taiwan, funting.huang@gmail.com)

It has been anecdotally observed that the three-way contrasts in Standard Chinese are reduced to two-way contrasts in Penang Mandarin (PM). PM is a variety of Mandarin Chinese spoken in Penang of Malaysia, which is influenced by Penang Hokkien. This work shows that the alleged neutralization of contrasts is incomplete (10 consonants x 3 vowel contexts x 5 speakers). More specifically, alveopalatal [ɕ] may range from postalveolar zone (73.33%) to alveolar zone (26.67%), and so does retroflex [ʂ] (46.67% vs. 46.67%). [s] and [n] are apical (or [+anterior]) coronals. The goal of this study is three-fold: (i) to describe the places of articulation of PM coronals and the patterns of ongoing sound changes, (ii) to show the neutralization of place contrasts is incomplete whereby constriction length remains distinct for these sibilant sounds, and (iii) to demonstrate different coarticulatory patterns of consonants in variant vowel contexts. The intricate division of coronal consonants does not warrant a precise constriction location on the upper palate. This PM data lend support to Ladefoged and Wu's (1984) observation that it is not easy to pin down a clear-cut boundary between dental and alveolar stops, and between alveolar and palatoalveolar fricatives.

2pSC8. Final voicing and devoicing in American English. Olga Dmitrieva (Linguistics/School of Lang. and Cultures, Purdue Univ., 100 North University St., Beering Hall, Rm. 1289, West Lafayette, IN 47907, odmitrie@purdue.edu)

English is typically described as a language in which voicing contrast is not neutralized in word-final position. However, a tendency towards devoicing (at least partial) of final voiced obstruents in English has been reported by the previous studies (e.g., Docherty (1992) and references therein). In the present study, we examine a number of acoustic correlates of obstruent voicing and the robustness with which each one is able to differentiate between voiced and voiceless obstruents in the word-final position in the speech recorded by twenty native speakers of the Mid-Western dialect of American English. The examined acoustic properties include preceding vowel duration, closure or frication duration, duration of the release portion, and duration of voicing during the obstruent closure, frication, and release. Initial results indicate that final voiced obstruents are significantly different from the voiceless ones in terms of preceding vowel duration and closure/frication duration. However, release duration for stops does not appear to correlate with voicing in an equally reliable fashion. A well-pronounced difference in terms of closure voicing between voiced and voiceless final stops is significantly reduced in fricative consonants, which indicates a tendency towards neutralization of this particular correlate of voicing in the word-final fricatives of American English.

2pSC9. An analysis of the singleton-geminate contrast in Japanese fricatives and stops. Christopher S. Rourke and Zack Jones (Linguist, The Ohio State Univ., 187 Clinton St., Columbus, OH 43202, rourke.16@osu.edu)

Previous acoustic analyses of the singleton-geminate contrast in Japanese have focused primarily on read speech. The present study instead analyzed the lengths of singleton and geminate productions of word-medial fricatives and voiceless stops in spontaneous monologues from the Corpus of Spontaneous Japanese (Maekawa, 2003). The results of a linear mixed effects regression model mirrored previous findings in read speech that the geminate effect (the durational difference between geminate and singletons) of stops is significantly larger than that of fricatives. This study also found a large range of variability in the geminate effect size between talkers. The size of the geminate effect between fricatives and voiceless stops was found to be slightly correlated, suggesting that they might be related to other rate-associated production differences between individuals. This suggestion was evaluated by exploring duration differences associated with talker age and gender. While there was no relationship between age and duration, males produced shorter durations than females for both fricatives and stops. However, the size of the geminate effect was not related to the gender of the speaker. The cause of these individual differences may be related to sound perception. Future research will investigate the cause of these individual differences in geminate effect size.

2pSC10. Quantifying surface phonetic variation using acoustic landmarks as feature cues. Jeung-Yoon Choi and Stefanie Shattuck-Hufnagel (Res. Lab. of Electronics, MIT, 50 Vassar St., Rm. 36-523, Cambridge, MA 02139, sshuf@mit.edu)

Acoustic landmarks, which are abrupt spectral changes associated with certain feature sequences in spoken utterances, are highly informative and have been proposed as the initial analysis stage in human speech perception, and for automatic speech recognition (Stevens, JASA 111(4), 2002, 1872–1891). These feature cues and their parameter values also provide an effective tool for quantifying systematic context-governed surface phonetic variation (Shattuck-Hufnagel and Veilleux, ICPhS XVI, 2007, 925–928). However, few studies have provided landmark-based information about the full range of variation in continuous communicative speech. The current study examines landmark modification patterns in a corpus of maptask-elicited speech, hand annotated for whether the landmarks were realized as predicted from the word forms or modified in context. Preliminary analyses of a single conversation (400 s, one speaker) show that the majority of landmarks (about 84%) exhibited the canonical form predicted from their lexical specifications, and that modifications were distributed systematically across segment types. For example, 90% of vowel landmarks (at amplitude/F1 peaks) were realized as predicted, but only 70% of the closures for non-strident fricatives /v/ and /dh/, and 40–50% of /t/ closures and releases. Further quantification of landmark modification patterns will provide useful information about the processing of surface phonetic variation.

2pSC11. Age- and gender-related variation in voiced stop prenasalization in Japanese. Mieko Takada (Aichi Gakuin Univ., Nisshin, Japan), Eun Jong Kong (Korea Aerosp. Univ., Goyang-City, South Korea), Kiyoko Yoneyama (Daito Bunka Univ., 1-9-1 Takashimadaira, Itabashi-ku, Tokyo 175-8571, Japan, yoneyama@ic.daito.ac.jp), and Mary E. Beckman (Ohio State Univ., Columbus, OH)

Modern Japanese is generally described as having phonologically voiced (versus voiceless) word-initial stops. However, phonetic details vary across dialects and age groups; in Takada's (2011) measurements of recordings of 456 talkers from multiple generations of talkers across five dialects, Osaka-area speakers and older speakers in the Tokyo area (Tokyo, Chiba, Saitama, and Kanagawa prefectures) typically show pre-voicing (lead VOT), but younger speakers show many "devoiced" (short lag VOT) values, a tendency that is especially pronounced among younger Tokyo-area females. There is also variation in the duration of the voice bar, with very long values (up to -200 ms lead VOT) observed in the oldest female speakers. Spectrograms of such tokens show faint formants during the stop closure, suggesting a velum-lowering gesture to vent supra-glottal air pressure to sustain vocal fold vibration. Further evidence of pre-nasalization in older Tokyo-area females comes from comparing amplitude trajectories for the voice bar to amplitude trajectories during nasal consonants, adapting a method proposed by Burton, Blumstein, and Stevens (1972) for exploring phonemic pre-nasalization contrasts. Differences in trajectory shape patterns between the oldest males and females and between older and younger females are like the differences that Kong, Syrika, and Edwards (2012) observed across Greek dialects.

2pSC12. An acoustic comparison of dental and retroflex sibilants in Chinese Mandarin and Taiwan Mandarin. Hanbo Yan and Allard Jongman (Linguist, Univ. of Kansas, 1732 Anna Dr., Apt. 11, Lawrence, KS 66044, yanhanbo@ku.edu)

Mandarin has both dental and retroflex sibilants. While the Mandarin varieties spoken in China and Taiwan are often considered the same, native speakers of Mandarin can tell the difference between the two. One obvious difference is that between the retroflex ([ʂ], [tʂ], [tʂh]) and dental sibilants ([s], [ts], [tsh]). This study investigates the acoustic properties of the sibilants of Chinese Mandarin and Taiwan Mandarin. Eight native speakers each of Chinese and Taiwan Mandarin produced the six target sibilants in word-initial position. A number of acoustic parameters, including spectral moments and duration, were analyzed to address two research questions: (a) which parameters distinguish the dental and retroflex in each type of Mandarin; (b) is there a difference between Chinese and Taiwan Mandarin? Results show that retroflex sibilants have a lower M1 and M2, and a higher M3 than dental sibilants in each language. Moreover, Chinese Mandarin has significantly larger M1, M2, and M3 differences than Taiwan Mandarin. This pattern suggests that, in contrast to Chinese Mandarin, Taiwan Mandarin is merging the retroflex sibilants in a dental direction.

2pSC13. Statistical relationships between phonological categories and acoustic-phonetic properties of Korean consonants. Noah H. Silbert (Commun. Sci. & Disord., Univ. of Cincinnati, 3202 Eden Ave., 344 French East Bldg., Cincinnati, OH 45267, noah.silbert@uc.edu) and Hanyong Park (Linguist, Univ. of Wisconsin, Milwaukee, WI)

The mapping between segmental contrasts and acoustic-phonetic properties is complex and many-to-many. Contrasts are often cued by a multiple acoustic-phonetic properties, and acoustic-phonetic properties typically provide information about multiple contrasts. Following the approach of de Jong *et al.* (2011, JASA 129, 2455), we analyze multiple native speakers' repeated productions of Korean obstruents using a hierarchical multivariate statistical model of the relationship between multidimensional acoustics and phonological categories. Specifically, we model the mapping between categories and multidimensional acoustic measurements from multiple repetitions of 14 Korean obstruent consonants produced by 20 native speakers (10 male, 10 female) in onset position in monosyllables. The statistical model allows us to analyze distinct within- and between-speaker sources of variability in consonant production, and model comparisons allow us to assess the utility of complexity in the assumed underlying phonological category system. In addition, by using the same set of acoustic measurements for the current project's Korean consonants and the English consonants analyzed by de Jong *et al*., we can model the within- and between-language acoustic similarity of phonological categories, providing a quantitative basis for predictions about cross-language phonetic perception.

2pSC14. Corpus testing a fricative discriminator: Or, just how invariant is this invariant? Philip J. Roberts (Faculty of Linguist, Univ. of Oxford, Ctr. for Linguist and Philology, Walton St., Oxford OX1 2HG, United Kingdom, philip.roberts@ling-phil.ox.ac.uk), Henning Reetz (Institut fuer Phonetik, Goethe-Universitaet Frankfurt, Frankfurt-am-Main, Germany), and Aditi Lahiri (Faculty of Linguist, Univ. of Oxford, Oxford, United Kingdom)

Acoustic cues to the distinction between sibilant fricatives are claimed to be invariant across languages. Evers *et al*. (1998) present a method for distinguishing automatically between [s] and [ʃ], using the slope of regression lines over separate frequency ranges within a DFT spectrum. They report accuracy rates in excess of 90% for fricatives extracted from recordings of minimal pairs in English, Dutch and Bengali. These findings are broadly replicated by Maniwa *et al*. (2009), using VCV tokens recorded in the lab. We tested the algorithm from Evers *et al*. (1998) against tokens of fricatives extracted from the TIMIT corpus of American English read speech, and the Kiel corpora of German. We were able to achieve similar accuracy rates to those reported in previous studies, with the following caveats: (1) the measure relies on being able to perform a DFT for frequencies from 0 to 8 kHz, so that a minimum sampling rate of 16 kHz is necessary for it to be effective, and (2) although the measure draws a similarly clear distinction between [s] and [ʃ] to those found in previous studies, the threshold value between the two sounds is sensitive to the dynamic range of the input signal.

2pSC15. Discriminant variables for plosive- and fricative-type single and geminate stops in Japanese. Shigeaki Amano and Kimiko Yamakawa (Faculty of Human Informatics, Aichi Shukutoku Univ., 2-9 Katahira, Nagakute, Aichi 480-1197, Japan, psy@asu.aasa.ac.jp)

Previous studies suggested that a plosive-type geminate stop in Japanese is discriminated from a single stop with variables of stop closure duration and subword duration that spans from the mora preceding the geminate stop to the vowel following the stop. However, this suggestion does not apply to a fricative-type geminate stop that does not have a stop closure. To overcome this problem, this study proposes Inter-Vowel Interval (IVI) and Successive Vowel Interval (SVI) as discriminant variables. IVI is the duration between the end of the vowel preceding the stop and the beginning of the vowel following the stop. SVI is the duration between the beginning of the vowel preceding the stop and the end of the vowel following the stop. When discriminant analysis was conducted between single and geminate stops of plosive and fricative types using IVI and SVI as independent variables, the discriminant ratio was very high (99.5%, n = 368). This result indicates that IVI and SVI are the general variables that represent acoustic features distinguishing Japanese single and geminate stops of both plosive and fricative types. [This study was supported by JSPS KAKENHI Grant Numbers 24652087, 25284080, 26370464 and by Aichi-Shukutoku University Cooperative Research Grant 2013-2014.]

2pSC16. Perceptual affinity of Mandarin palatals and retroflexes. Yunghsiang Shawn Chang (Dept. of English, National Taipei Univ. of Technol., Zhongxiao E. Rd., Sec. 3, No. 1, Taipei 106, Taiwan, shawnchang@ntut.edu.tw)

Mandarin palatals [tɕ, tɕ^h, ɕ], which only occur before [i, y] vowels, are in complementary distribution with the alveolars [ts, ts^h, s], the velars [k, k^h, x], and the retroflexes [tʂ, tʂ^h, ʂ]. Upon investigating perceptually motivated accounts for the phonological representation of the palatals, Wan (2010) reported that Mandarin palatals were considered more similar to the alveolars than the velars, whereas Lu (2014) found categorical results for the palatal-alveolar discrimination. The current study furthered the investigation to the perceptual affinity between Mandarin palatals and retroflexes by having 15 native listeners identify two 8-step resynthesized [ʂ-s] continua (adapted from Chang *et al*. (2013)) cross-spliced with [i, y] vowels, respectively. To avoid phonotactic restrictions from biasing perception, all listeners were trained on identifying the [ɕi, ʂi, si] and [ɕy, ʂy, sy] syllables produced by a phonetician before the experiment. The results showed that all resynthesized stimuli, though lacking palatal-appropriate vocalic transitions, were subject to palatal perception. Particularly, two intermediate steps along the [ʂi-si] continuum and five along the [ʂy-sy] continuum were identified as palatal syllables by over 70% of the listeners. The results suggest that Mandarin palatals could be identified with both the retroflexes and alveolars based on perceptual affinity.

2pSC17. Perceptual distinctiveness of dental vs. palatal sibilants in different vowel contexts. Mingxing Li (Linguist, The Univ. of Kansas, 1541 Lilac Ln., Blake Hall 427, Lawrence, KS 66045-3129, mxlistar@gmail.com)

This paper reports a similarity rating experiment and a speeded AX discrimination experiment to test the perceptual distinctiveness of dental vs. palatal sibilants in different vowel contexts. The stimuli were pairs of CV sequences where the onsets were [s, ts, ts^h] vs. [ɕ, tɕ, tɕ^h] as in Mandarin Chinese and the vowels were [i, a, o]; the durations of the consonants and vowels were set to values close to those in natural speech; the inter-stimulus-interval was set at 100ms to facilitate responses based on psychoacoustic similarity. A significant effect of vowel contexts was observed in the similarity rating by 20 native American English speakers, whereby the dental vs. palatal sibilants were judged to be the least distinct in the [i] context. A similar pattern was observed in the speeded AX discrimination, whereby the [i] context introduced slower "different" responses than other vowels. In general, this study supports the view that the perceptual distinctiveness of a consonant pair may vary with different vowel contexts. Moreover, the experiment results match the typological pattern of dental vs. palatal sibilants across Chinese dialects, where contrasts like /si, tsi, ts^{h}i/ vs. /ɕi, tɕi, tɕ^{h}i/ are often enhanced by vowel allophony.

2pSC18. Phonetic correlates of stance-taking. Valerie Freeman, Richard Wright, Gina-Anne Levow (Linguist, Univ. of Washington, Box 352425, Seattle, WA 98195, valerief@uw.edu), Yi Luan (Elec. Eng., Univ. of Washington, Seattle, WA), Julian Chan (Linguist, Univ. of Washington, Seattle, WA), Trang Tran, Victoria Zayats (Elec. Eng., Univ. of Washington, Seattle, WA), Maria Antoniak (Linguist, Univ. of Washington, Seattle, WA), and Mari Ostendorf (Elec. Eng., Univ. of Washington, Seattle, WA)

Stance, or a speaker's attitudes or opinions about the topic of discussion, has been investigated textually in conversation- and discourse analysis and in computational models, but little work has focused on its acoustic-phonetic properties. This is a difficult problem, given that stance is a complex activity that must be expressed along with several other types of meaning (informational, social, etc.) using the same acoustic channels. In this presentation, we begin to identify some acoustic indicators of stance in natural speech using a corpus of collaborative conversational tasks which have been hand-annotated for stance strength (none, weak, moderate, and strong) and polarity (positive, negative, and neutral). A preliminary analysis of 18 dyads completing two tasks suggests that increases in stance strength are correlated with increases in speech rate and pitch and intensity medians and ranges. Initial results for polarity also suggest correlations with speech rate and intensity. Current investigations center on local modulations in pitch and intensity, durational and spectral differences between stressed and unstressed vowels, and disfluency rates in different stance conditions. Consistent male/female differences are not yet apparent but will also be examined further.

2pSC19. Compounds in modern Greek. Angeliki Athanasopoulou and Irene Vogel (Linguist and Cognit. Sci., Univ. of Delaware, 125 East Main St., Newark, DE 19716, angeliki@udel.edu)

The difference between compounds and phrases has been studied extensively in English (e.g., Farnetani, Torsello, & Cosi, 1988; Plag, 2006; Štekauer, Zimmermann, & Gregová, 2007). However, little is known about the analogous difference in Modern Greek (Tzakosta, 2009). Greek compounds (Ralli, 2003) form a single phonological word, and thus, they only contain one primary stress. That means that the individual words lose their primary stress. The present study is the first acoustic investigation of the stress properties of Greek compounds and phrases. Native speakers of Greek produce ten novel adjective + noun compounds and their corresponding phrases (e.g., phrase: [kócino δódi] "a red tooth" vs. compound: [kocino$\delta\delta$ódis] "someone with red teeth") in the sentence corresponding to "The XXX is at the top/bottom of the screen." Preliminary results confirm the earlier descriptive claims that compounds only have a single stress, while phrases have one on each word. Specifically, the first word (i.e., adjective) in compounds is reduced in F0 (101 Hz), duration (55 ms), and intensity (64 dB) compared to phrases (F0 = 117Hz, duration = 85 ms, and intensity = 67 dB). Also, both words are very similar for all of the measures in phrases. The second word (i.e., noun) is longer than the first word, possibly indicating phrase-final lengthening.

2pSC20. Evoked potentials during voice error detection at register boundaries. Anjli Lodhavia (Dept. of Commun. Disord. and Sci., Rush Univ., 807 Reef Court, Wheeling, IL 60090, alodhavia1@gmail.com), Sona Patel (Dept. of Speech-Lang. Pathol., Seton Hall Univ., South Orange, NJ), Saul Frankford (Dept. of Commun. Sci. and Disord., Northwestern Univ., Tempe, Arizona), Oleg Korzyukov, and Charles R. Larson (Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Singers require great effort to avoid vocal distortion at register boundaries, as they are trained to diminish the prominence of register breaks. We examined neural mechanisms underlying voice error detection in singers at their register boundaries. We hypothesized that event-related potentials (ERPs), reflecting brain activity, would be larger if a singer's pitch was unexpectedly shifted toward, rather than away, from their register break. Nine trained singers sustained a musical note for ~3 seconds near their modal register boundaries. As the singers sustained these notes, they heard their voice over headphones shift in pitch (+/- 400 cents, 200 ms) either toward or away from the register boundary. This procedure was repeated for 200 trials. The N1 and P2 ERP amplitudes for three central electrodes (FCz, Cz, Fz) were computed from the EEGs of all participants. Results of a multivariate analysis of variance for shift direction (+400c, -400c) and register (low, high) showed significant differences in N1 and P2 amplitude for direction at the low boundary of modal register, but not the high register boundary. These results may suggest increased neural activity in singers when trying to control the voice when crossing the lower register boundary.

2pSC21. The articulatory tone-bearing unit: Gestural coordination of lexical tone in Thai. Robin P. Karlin and Sam Tilsen (Linguist, Cornell Univ., 103 W Yates St., Ithaca, NY 14850, karlin.robin@gmail.com)

Recently, tones have been analyzed as articulatory gestures that can coordinate with segmental gestures. In this paper, we show that the tone gestures that make up a HL contour tone are differentially coordinated with articulatory gestures in Thai syllables, and that the coordinative patterns are influenced by the segments and moraic structure of the syllables. The autosegmental approach to lexical tone describes tone as a suprasegment that must be associated to some tone-bearing unit (TBU); in Thai, the language of study, the proposed TBU is the mora. Although the autosegmental account largely describes the phonological patterning of tones, it remains unclear how the abstract representation of tone is implemented. An electromagnetic articulograph (EMA) study of four speakers of Thai was conducted to examine the effects of segment type and moraic structure on the coordination of tone gestures. In a HL contour tone, tone gestures behave similarly to consonant gestures, and show patterns of coordination with gestures that correspond to moraic segments. However, there is also a level of coordination between the H and L tone gestures. Based on these results, a model of TBUs is proposed within the Articulatory Phonology framework that incorporates tone-segment coordination as well as tone-tone coordination.

2pSC22. The role of prosody in English sentence disambiguation. Taylor L. Miller (Linguist & Cognit. Sci., Univ. of Delaware, 123 E Main St., Newark, DE 19716, tlmiller@udel.edu)

Only certain ambiguous sentences are perceptually disambiguable. Some researchers argue that this is due to syntactic structure (Lehiste 1973, Price 1991, Kang & Speer 2001), while others argue prosodic structure is responsible (Nespor & Vogel 1986 = N&V, Hirshberg & Avesani 2000). The present study further tests the role of prosodic constituents in sentence disambiguation in English. Target sentences were recorded in disambiguating contexts; twenty subjects listened to the recordings and chose one of two meanings. Following N&V's experimental design with Italian, the meanings of each target structurally corresponded to different syntactic constituents and varied with respect to phonological phrases (ϕ) and intonational phrases (I). The results confirm N&V's Italian findings: listeners are only able to disambiguate sentences with different prosodic constituent structures ($p < 0.05$); those differing in (I) but not (ϕ) have the highest success rate—86% (e.g., [When danger threatens your children]I [call the police]I vs. [When danger threatens]I [your children call the police]I). As reported elsewhere (e.g., Lehiste 1973), we also observed a meaning bias in some cases (e.g., in "Julie ordered some large knife sharpeners," listeners preferred "large [knife sharpeners]" but in "Jill owned some gold fish tanks," they preferred "[goldfish] tanks").

2pSC23. Perceptual isochrony and prominence in spontaneous speech. Tuuli Morrill (Linguist, George Mason Univ., 4400 University Dr., 3E4, Fairfax, VA, tmorrill@msu.edu), Laura Dilley (Commun. Sci. and Disord., Michigan State Univ., East Lansing, MI), and Hannah Forsythe (Linguist, Michigan State Univ., East Lansing, MI)

While it has been shown that stressed syllables do not necessarily occur at equal time intervals in speech (Cummins, 2005; Dauer, 1983), listeners frequently perceive stress as occurring regularly, a phenomenon termed perceptual isochrony (Lehiste, 1977). A number of studies have shown that in controlled experimental materials, a perceptually isochronous sequence of stressed syllables generates expectations which affect word segmentation and lexical access in subsequent speech (e.g., Dilley & McAuley, 2008). The present research used the Buckeye Corpus of Conversational Speech (Pitt *et al*., 2007) to address two main questions (1) What acoustic and linguistic factors are associated with the occurrence of perceptual isochrony? and (2) What are the effects of perceptually isochronous speech passages on the placement of prominence in subsequent speech? In particular, we investigate the relationship between perceptual isochrony and lexical items traditionally described as "unstressed" (e.g., grammatical function words), testing whether these words are more likely to be perceived and/or produced as prominent when they are preceded and/or followed by a perceptually isochronous passage. These findings will contribute to our understanding of the relationship between acoustic correlates of phrasal prosody and lexical perception. [Research partially supported by NSF CAREER Award BCS 0874653 to L. Dilley.]

2pSC24. French listeners' processing of prosodic focus. Jui Namjoshi (French, Univ. of Illinois at Urbana-Champaign, 2090 FLB, MC-158, S. Mathews Ave, Urbana, IL 61801, namjosh2@illinois.edu)

Focus in French, typically conveyed by syntax (e.g., clefting) with prosody, can be signaled by prosody alone (contrastive pitch accents on the first syllable of focused constituents, cf. nuclear pitch accents, on the last nonreduced syllable of the Accentual Phrase) (Féry, 2001; Jun & Fougeron, 2000). Do French listeners, like L1-English listeners (Ito & Speer, 2008) use contrastive accents to anticipate upcoming referents? 20 French listeners completed a visual-world eye-tracking experiment. Cross-spliced, amplitude-neutralized stimuli included context (1) and critical (2) sentences in a 2x2 design, with accent on object (nuclear/ contrastive) and person's information status (new/ given) as within-subject variables (see (1)-(2)). Average amplitudes and durations for object words were 67 dB and 0.68 s for contrastive accents, and 63.8 dB and 0.56 s for nuclear accents, respectively. Mixed-effects models showed a significant effect of accent-by-information-status interaction on competitor fixation proportions in the post-disambiguation time window ($p<0.05$). Contrastive accents yielded lower competitor fixation proportions with a given person than with a new person, suggesting that contrastive accents constrain lexical competition in French. (1) Clique sur le macaRON de Marie-Hélène. (2) Puis clique sur le chocoLAT/ CHOcolat de Marie-Hélène/ Jean-Sébastien. (nuclear/contrastive accent, given/new person) '(Then) Click on the macaron/chocolate of Marie-Hélène/Jean-Sébastien.'

2pSC25. Prominence, contrastive focus and information packaging in Ghanaian English discourse. Charlotte F. Lomotey (Texas A&M University-Commerce, 1818D Hunt St., Commerce, TX 75428, cefolatey@yahoo.com)

Contrastive focus refers to the coding of information that is contrary to the presuppositions of the interlocutor. Thus, in everyday speech, speakers employ prominence to mark contrastive focus such that it gives an alternative answer to an explicit or implicit statement provided by the previous discourse or situation (Rooth, 1992), and plays an important role in facilitating language understanding. Even though contrastive focus has been investigated in native varieties of English, there is little or no knowledge of similar studies as far as non-native varieties of English, including that of Ghana, are concerned. The present study investigates how contrastive focus is marked with prosodic prominence in Ghanaian English, and how such a combination creates understanding among users of this variety. To achieve this, data consisting of 61/2 hours of English conversations from 200 Ghanaians were analyzed using both auditory and acoustic means. Results suggest that Ghanaians tend to shift the contrastive focus from the supposed focused syllable onto the last syllable of the utterance, especially when that syllable ends the utterance. Although such tendencies may shift the focus of the utterance, the data suggest that listeners do not seem to have any problem with speakers' packaging of such information.

2pSC26. The representation of tone 3 sandhi in Mandarin: A psycholinguistic study. Yu-Fu Chien and Joan Sereno (Linguist, The Univ. of Kansas, 1407 W 7th St., Apt. 18, Lawrence, KS 66044-6716, whouselefthand@gmail.com)

In Mandarin, tone 3 sandhi is a tonal alternation phenomenon in which a tone 3 syllable changes to a tone 2 syllable when it is followed by another tone 3 syllable. Thus, the initial syllable of Mandarin bisyllabic sandhi words is tone 3 underlyingly but becomes tone 2 on the surface. An auditory-auditory priming lexical decision experiment was conducted to investigate how Mandarin tone 3 sandhi words are processed by Mandarin native listeners. The experiment examined prime-target pairs, with monosyllabic primes and bisyllabic Mandarin tone 3 sandhi targets. Each tone sandhi target word was preceded by one of three corresponding monosyllabic primes: a tone 2 prime (Surface-Tone overlap) (chu2-chu3li3), a tone 3 prime (Underlying-Tone overlap) (chu3-chu3li3), or a control prime (Baseline condition) (chu1-chu3li3). In order to assess the contribution of frequency of occurrence, 15 High Frequency and 15 Low Frequency sandhi target words were used. Thirty native speakers of Mandarin participated. Results showed that tone 3 sandhi targets elicited significantly stronger facilitation effects in the Underlying-Tone condition than in the Surface-Tone condition, with little effect of frequency of occurrence. The data will be discussed in terms of lexical access and the nature of the representation of Mandarin words.

2pSC27. Perception of sound symbolism in mimetic stimuli: The voicing contrast in Japanese and English. Kotoko N. Grass (Linguist, Univ. of Kansas, 9953 Larsen St., Overland Park, KS 66214, nakata.k@ku.edu) and Joan Sereno (Linguist, Univ. of Kansas, Lawrence, KS)

Sound symbolism is a concept in which the sound of a word and the meaning of the word are systematically related. The current study investigated whether the voicing contrast between voiced /d, g, z/ and voiceless /t, k, s/ consonants systematically affects categorization of Japanese mimetic stimuli along a number of perceptual and evaluative dimensions. For the nonword stimuli, voicing of consonants was also manipulated, creating a continuum from voiced to voiceless endpoints (e.g., [gede] to [kete]), in order to examine the categorical nature of the perception. Both Japanese native speakers and English native speakers, who had no knowledge of Japanese, were examined. Stimuli were evaluated on size (big–small) and shape (round–spiky) dimensions as well as two evaluative dimensions (good–bad, graceful–clumsy). In the current study, both Japanese and English listeners associated voiced sounds with largeness, badness, and clumsiness and voiceless sounds with smallness, goodness, and gracefulness. For the shape dimension, however, English and Japanese listeners showed contrastive categorization, with English speakers associating voiced stops with roundness and Japanese listeners associating voiced stops with spikiness. Interestingly, sound symbolism was very categorical in nature. Implications of the current data for theories of sound symbolism will be discussed.

Session 2pUW

Underwater Acoustics: Propagation and Scattering

Megan S. Ballard, Chair
Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, TX 78758

Contributed Papers

1:00

2pUW1. Low frequency propagation experiments in Currituck Sound. Richard D. Costley (GeoTech. and Structures Lab., U.S. Army Engineer Res. & Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, dan.costley@usace.army.mil), Kent K. Hathaway (Coastal & Hydraulics Lab., US Army Engineer Res. & Development Ctr., DC, NC), Andrew McNeese, Thomas G. Muir (Appl. Res. Lab., Univ. of Texas at Austin, Austin, TX), Eric Smith (GeoTech. and Structures Lab., U.S. Army Engineer Res. & Development Ctr., Vicksburg, Texas), and Luis De Jesus Diaz (GeoTech. and Structures Lab., U.S. Army Engineer Res. & Development Ctr., Vicksburg, MS)

In water depths on the order of a wavelength, sound propagates with considerable involvement of the bottom, whose velocities and attenuation vary with depth into the sediment. In order to study propagation in these types of environments, experiments were conducted in Currituck Sound on the Outer Banks of North Carolina using a Combustive Sound Source (CSS) and bottom mounted hydrophones and geophones as receivers. The CSS was deployed at a depth of approximately 1 meter and generated transient signals, several wavelengths long, at frequencies around 300 Hz. The results are used to determine transmission loss in water depths of approximately 3 meters, as well as to examine the generation and propagation of Sholte type interface waves. The measurements are compared to numerical models generated with a two-dimensional finite-element code. [Work supported by the U.S. Army Engineer Research and Development Center. Permission to publish was granted by Director, Geotechnical & Structures Laboratory.]

1:15

2pUW2. Three-dimensional acoustic propagation effect in subaqueous sand dune field. Andrea Y. Chang, Chi-Fang Chen (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, yychang@ntu.edu.tw), Linus Y. Chiu (Inst. of Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan), Emily Liu (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan), Ching-Sang Chiu, and Davis B. Reeder (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA)

Very large subaqueous sand dunes are discovered on the upper continental slope of the northern South China Sea in water depth of 160–600 m, which composed of fine to medium sand. The amplitude and the crest-to-crest wavelength of sand dunes are about 5–15 m and 200–400 m, respectively. This topographic feature could causes strong acoustic scattering, mode coupling, and out-of- plane propagation effects, which consequently result in sound energy redistribution within ocean waveguide. This research focus on the three-dimensional propagation effects (e.g., horizontal refraction) induced by the sand dunes in the South China Sea, which are expected as the angle of propagation relative to the bedform crests decreases. The three-dimensional propagation effects are studied by numerical modeling and model-data comparison. For numerical modeling, the *in-situ* topographic data of subaqueous sand dune and sound speed profiles were inputted to calculate the acoustic fields, which were further decomposed into mode fields to show the modal horizontal refraction effects. The modeling results were manifested by data observations. [This work is sponsored by the Ministry of Science and Technology of Taiwan.]

1:30

2pUW3. Results from a scale model acoustic propagation experiment over a translationally invariant wedge. Jason D. Sagers (Environ. Sci. Lab., Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu)

A 1:7500 scale underwater acoustic propagation experiment was conducted in a laboratory tank to investigate three-dimensional (3D) propagation effects, with the objective of providing benchmark quality data for comparison with numerical models. A computer controlled positioning system accurately moves the receiving hydrophone in 3D space while a stationary source hydrophone emits band-limited pulse waveforms between 200 kHz and 1 MHz. The received time series can be post-processed to estimate travel time, transmission loss, and vertical and horizontal arrival angle. Experimental results are shown for a 1.22×2.13 m bathymetric part possessing both a flat bottom bathymetry and a translationally invariant wedge with a 10° slope. Comparisons between the experimental data and numerical models are also shown. [Work supported by ONR.]

1:45

2pUW4. Numerical modeling of measurements from an underwater scale-model tank experiment. Megan S. Ballard and Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

Scale-model tank experiments are beneficial because they offer a controlled environment in which to make underwater acoustic propagation measurements, which is helpful when comparing measured data to calculations from numerical propagation models. However, to produce agreement with the measured data, experimental details must be carefully included in the model. For example, the frequency-dependent transmitting and receiving sensitivity and vertical directionality of both hydrophones must be included. In addition, although it is possible to measure the geometry of the tank experiment, including water depth and source and receiver positions, positional uncertainty exists due to the finite resolution of the measurements. The propagated waveforms from the experiment can be used to resolve these parameters using inversion techniques. In this talk, model-data comparisons of measurements made in a 1:7500 scale experiment are presented. The steps taken to produce agreement between the measured and modeled data are discussed in detail for both range-independent and range-dependent configurations.

2:00

2pUW5. A normal mode inner product to account for acoustic propagation over horizontally variable bathymetry. Charles E. White, Cathy Ann Clark (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, charlie.e.white@navy.mil), Gopu Potty, and James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

This talk will consider the conversion of normal mode functions over local variations in bathymetry. Mode conversions are accomplished through an inner product, which enables the modes compromising the field at each range-dependent step to be written as a function of those in the preceding step. The efficiency of the method results from maintaining a stable number

of modes throughout the calculation of the acoustic field. A verification of the inner product is presented by comparing results from its implementation in a simple mode model to that of a closed-form solution for the acoustic wedge environment. A solution to the more general problem of variable bottom slope, which involves a decomposition of bathymetric profiles into a sequence of wedge environments, will also be discussed. The overall goal of this research is the development and implementation of a rigorous shallow water acoustic propagation solution which executes in a time window to support tactical applications.

2:15

2pUW6. An assessment of the effective density fluid model for backscattering from rough poroelastic interfaces. Anthony L. Bonomo, Nicholas P. Chotiros, and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

The effective density fluid model (EDFM) was developed to approximate the behavior of sediments governed by Biot's theory of poroelasticity. Previously, it has been shown that the EDFM predicts reflection coefficients and backscattering strengths that are in close agreement with those of the full Biot model for the case of a homogeneous poroelastic half-space. However, it has not yet been determined to what extent the EDFM can be used in place of the full Biot model for other cases. In this work, the finite element method is used to compare the backscattering strengths predicted using the EDFM with the predictions of the full Biot model for three cases: a homogeneous poroelastic half-space with a rough interface, a poroelastic layer overlying an elastic half-space with both interfaces rough, and an inhomogeneous poroelastic half-space consisting of a shear modulus gradient with a rough interface. [Work supported by ONR, Ocean Acoustics.]

2:30

2pUW7. Scattering by randomly rough surfaces. I. Analysis of slope approximations. Patrick J. Welton (Appl. Res. Lab., The Univ. of Texas at Austin, 1678 Amarelle St., Thousand Oaks, CA 91320-5971, patrickwelton@verizon.net)

Progress in numerical methods now allows scattering in two dimensions to be computed without resort to approximations. However, scattering by three-dimensional random surfaces is still beyond the reach of current numerical techniques. Within the restriction of the Kirchhoff approximation (single scattering) some common approximations used to predict scattering by randomly rough surfaces will be examined. In this paper, two widely used approximate treatments for the surface slopes will be evaluated and compared to the exact slope treatment.

2:45–3:00 Break

3:00

2pUW8. Scattering by randomly rough surfaces. II. Spatial spectra approximations. Patrick J. Welton (Appl. Res. Lab., The Univ. of Texas at Austin, 1678 Amarelle St., Thousand Oaks, CA 91320-5971, patrickwelton@verizon.net)

The spatial spectrum describing a randomly rough surface is crucial to the theoretical analysis of the scattering behavior of the surface. Most of the models assume that the surface displacements are a zero-mean process. It is shown that a zero-mean process requires that the spatial spectrum vanish when the wavenumber is zero. Many of the spatial spectra models used in the literature do not meet this requirement. The impact of the zero-mean requirement on scattering predictions will be discussed, and some spectra models that meet the requirement will be presented.

3:15

2pUW9. Scattering by randomly rough surfaces. III. Phase approximations. Patrick J. Welton (Appl. Res. Lab., The Univ. of Texas at Austin, 1678 Amarelle St., Thousand Oaks, CA 91320-5971, patrickwelton@verizon.net)

In the limit as the roughness vanishes, the solution for the pressure scattered by a rough surface of infinite extent should reduce to the image solution. Approximate image solutions for an infinite, pressure-release plane surface are studied for an omnidirectional source using the 2nd, 3rd, and 4th order phase approximations. The results are compared to the exact image solution to examine the effects of the phase approximations. The result based on the 2nd order (Fresnel phase) approximation reproduces the image solution for all geometries. Surprisingly, the results for the 3rd and 4th order phase approximations are never better than the Fresnel result, and are substantially worse for most geometries. This anomalous behavior is investigated and the cause is found to be the multiple stationary phase points produced by the 3rd and 4th order phase approximations.

3:30

2pUW10. Role of binding energy (edge-to-face contact of mineral platelets) in the acoustical properties of oceanic mud sediments. Allan D. Pierce (Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net) and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

A theory for mud sediments presumes a card-house model, where the platelets arrange themselves in a highly porous configuration; electrostatic forces prevent face-to-face contacts. The primary type of contact is where the edge of one platelet touches a face of another. Why such is not also prevented by electrostatic forces is because of van der Waals (vdW) forces between the molecular structures within the two platelets. A quantitative assessment is given of such forces, taking into account the atomic composition and crystalline structure of the platelets, proceeding from the London theory of interaction between non-polar molecules. Double-integration over both platelets leads to a quantitative and simple prediction for the potential energy of vdW interaction as a function of the separation distance, edge-from-face. At moderate nanoscale distances, the resulting force is attractive and is much larger than the electrostatic repulsion force. But, at very close (touching) distances, the intermolecular force becomes also repulsive, so that there is a minimum potential energy, which is identified as the binding energy. This finite binding energy, given a finite environmental temperature, leads to some statistical mechanical theoretical implications. Among the acoustical implications is a relaxation mechanism for the attenuation of acoustic waves propagating through mud.

3:45

2pUW11. Near bottom self-calibrated measurement of normal reflection coefficients by an integrated deep-towed camera/acoustical system. Linus Chiu, Chau-Chang Wang, Hsin-Hung Chen (Inst. of Undersea Technol., National Sun Yat-sen Univ., No. 70, Lienhai Rd., Kaohsiung 80424, Taiwan, linus@mail.nsysu.edu.tw), Andrea Y. Chang (Asia-Pacific Ocean Res. Ctr., National Sun Yat-sen Univ., Kaohsiung, Taiwan), and Chung-Ray Chu (Inst. of Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan)

Normal incidence echo data (bottom reflection) can provide acoustic reflectivity estimates used to predict sediment properties with using seabed sediment models. Accuracy of normal reflection coefficient measurement thus become very significant to the bottom inversion result. A deep-towed camera platform with acoustical recording system, developed by the Institution of Undersea Technology, National Sun Yat-sen University, Taiwan, is capable of photographically surveying the seafloor in near scope and acquiring sound data. The real time data transference, including photography (optics) and reflection measurement (acoustics) can be implemented in the same site simultaneously. The deep-towed camera near the bottom was used in several experiments in the southwestern sea off Taiwan in 2014 to acquire acoustic LFM signal sent by surface shipboard source as incident signal as well as the seafloor reflections at frequency bands within 4–6 kHz. The error produced by compensating the roll-off of altitude of vehicle (propagation loss) can be eliminated, which is considered as near bottom self-calibrated measurement for normal reflection coefficient. The collected reflection coefficients were used to inverting the sediment properties with using the Effective Density Fluid model (EDFM), manifested by the coring and camera images. [This work is sponsored by the Ministry of Science and Technology of Taiwan.]

4:00

2pUW12. Backscattering from an obstacle immersed in an oceanic waveguide covered with ice. Natalie S. Grigorieva (St. Petersburg State Electrotech. Univ., 5 Prof. Popova Str., St. Petersburg 197376, Russian Federation, nsgrig@natalie.spb.su), Daria A. Mikhaylova, and Dmitriy B. Ostrovskiy (JSC "Concern Oceanpribor", St. Petersburg, Russian Federation)

The presentation describes the theory and implementation issues of modeling of the backscattering from an obstacle immersed in a homogeneous, range-independent waveguide covered with ice. An obstacle is assumed to be spherical, rigid or fluid body. A bottom of the waveguide and an ice cover are fluid, attenuating half-space. The properties of an ice cover and a scatterer may coincide. To calculate the scattering coefficients of a sphere [R. M. Hackman *et al*., J. Acoust. Soc. Am. 84, 1813–1825 (1988)], the normal mode evaluation is applied. A number of normal modes forming the backscattering field is determined by a given directivity of the source. The obtained analytical expression for the backscattered field is applied to evaluate its dependence on source frequency, depth of a water layer, bottom and ice properties, and distance between the source and obstacle. Two cases are analyzed and compared: when the upper boundary of a waveguide is sound-soft and when a water layer is covered with ice. Computational results are obtained in a wide frequency range 8–12 kHz for conditions of a shallow water testing area. [Work supported by Russian Ministry of Educ. and Sci., Grant 02.G25.31.0058.]

4:15

2pUW13. Emergence of striation patterns in acoustic signals reflected from dynamic surface waves. Youngmin Choo, Woojae Seong (Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul, Seoul 151 - 744, South Korea, sks655@snu.ac.kr), and Heechun Song (Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA)

A striation pattern can emerge in high-frequency acoustic signals interacting with dynamic surface waves. The striation pattern is analyzed using a ray tracing algorithm for both a sinusoidal and a rough surface. With a source or receiver close to the surface, it is found that part of the surface on either side of the specular reflection point can be illuminated by rays, resulting in time-varying later arrivals in channel impulse response that form the striation pattern. In contrast to wave focusing associated with surface wave crests, the striation occurs due to reflection off convex sections around troughs. Simulations with a sinusoidal surface show both an upward (advancing) and downward (retreating) striation patterns that depend on the surface-wave traveling direction and the location of the illuminated area. In addition, the striation length is determined mainly by the depth of the source or receiver, whichever is closer in range to the illuminated region. Even with a rough surface, the striation emerges in both directions. However, broadband (7–13 kHz) simulations in shallow water indicate that the longer striation in one direction is likely pronounced against a quiet noise background, as observed from at-sea experimental data. The simulation is extended for various surface wave spectra and it shows consistent patterns.

TUESDAY EVENING, 28 OCTOBER 2014 — 8:00 P.M. TO 9:30 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. On Tuesday the meetings will begin at 8:00 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:30 p.m. On Thursday evening, the meetings will begin at 7:30 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.)	Santa Fe
Acoustical Oceanography	Indiana G
Architectural Acoustics	Marriott 7/8
Physical Acoustics	Indiana C/D
Speech Communication	Marriott 3/4
Structural Acoustics and Vibration	Marriott 1/2