Session 3aAAa

Architectural Acoustics and Musical Acoustics: Directivities of Musical Instruments and Their Effects in Performance Environments, Room Simulations, Acoustical Measurements, and Audio II

Timothy W. Leishman, Chair
Physics and Astronomy, Brigham Young University, N247 ESC, Provo, UT 84602

Invited Papers

8:00

3aAAa1. Directivities of musical instruments that relate to their effects in performance environments and room simulations. Larry Kirkegaard (Kirkegaard Assoc., 801West Adams St., Chicago, IL 60607, lkirkegaard@kirkegaard.com)

This paper addresses the particular aspects of Directivities of Musical Instruments that relate to “Their Effects in Performance Environments and Room Simulations.” Large ensemble—small ensemble—soloist Flat floor—raked floor—risers—steep risers Wrap of the ensemble Relation to Soloists Amplified or acoustic Seated vs. Standing Effects for performers and effects for Audiences with various audience configurations. This paper is meant to be a celebration of new techniques combined with cautionary tales that speak to needs for “measuring equipment” that includes experienced ears and questioning minds engaged in designing the testing techniques and evaluating their results to extract the most pertinent and applicable information.

8:20

3aAAa2. Acoustical design of performance and rehearsal spaces influenced by instrument directivity. Peter D’Antonio (RPG Diffusor Systems, Inc., 651C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

The sound that we hear in a music space is determined by the direct sound, as well as the early and late reflections from the room’s surfaces, so instrument directivity and reflection control are of primary interest. Spatially directed sound by individual instruments and musical ensembles is controlled by the selection and placement of absorptive, reflective, and diffusive boundary surfaces, which contribute to the room’s preference for both the musicians and the audience. We will describe how these surfaces are measured and characterized for directional and random incidence and how they have been used in many types of musical environments. In this presentation, we will discuss and present before and after measurements in the design of an individual practice room addressing room modes; describe the essential ingredients in an ensemble rehearsal space; present before and after perception questionnaires for a variable acoustics modular performance shell for small music ensembles and a symphony orchestra; describe a stage canopy optimization as a function of the support objective measure and present computer model coverage predictions addressing rear wall and overhead canopy treatments in an auditorium.

8:40

3aAAa3. Using multi-channel anechoic recordings to represent source directivity in room acoustics models to improve auralizations. Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, vigeant@engr.psu.edu), Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Omaha, NE), and Jens Holger Rindel (Odeon A/S, Lyngby, Denmark)

A series of studies were conducted to investigate the effects of different sound source directivity representations on perceived differences in auralizations. The first study (Wang & Vigeant 2008) showed that subjects could not distinguish between auralizations created using both omnidirectional and measured static directivities of individual instruments. Differences may not have been observed since source directivities were not available for all octave bands of interest (125–8000 Hz). Significant differences were perceived between the auralizations generated using an omnidirectional source and a highly directional source for all bands. For the second study (Vigeant, Wang, & Rindel 2011), the effects of using 4- and 13-channel solo instrument anechoic recordings versus single channel recordings on realism and source width were evaluated. In general, the results showed an improvement in realism and decrease in source width with an increasing number of channels. In the final study (Vigeant, Wang, & Rindel 2008), 5-channel solo instrument anechoic recordings for two orchestral pieces were used to investigate the effects of using multi-channel and multi-source representations of an orchestra on realism, source depth, source width, and ease of distinguishing between instrument parts. For some of the cases studied, significant differences were found with the most complex representation of the orchestra.

9:00

3aAAa4. Quantification of time varying directivity of musical instruments in an orchestral context. Madeline A. Davidson and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska- Lincoln, 6349 Cedar Plaza #222, Omaha, NE 68106, madeline.davidson@huskers.unl.edu)

At the University of Nebraska, previous research has been done in the quantification of time varying directivity of played musical instruments using 13 channel anechoic recordings of solo passages. The proposed quantifiers involve measures of the extent to which directivity changes in magnitude as well as how often the directivity changes in spatial direction, using various time windows of analysis from 0.2 to 1 s. In this paper, these proposed quantifiers are applied to 5 channel anechoic recordings of a full orchestra for a Brahms symphony and a Mozart symphony. With silent sections removed from the recordings, the time varying directivity properties of musical instruments are more accurately examined within the context of played notes from two stylistically different symphonies.

9:20

3aAAa5. Interactive room auralization of sound sources with exchangeable directivities. Lukas Aspöck and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen, Kopernikusstr. 5, Aachen D-52074, Germany, las@akustik.rwth-aachen.de)

Although established measurement methods and simulation models for source directivities are available, there is a lack of tools to directly experience the audible effects of virtual sound sources in different conditions. Only an auralization in connection with psychoacoustic experiments can answer questions regarding the required solution and quality of sound source directivities. This paper presents a real-time simulation engine based on geometrical acoustics which accounts for both source directivities and the room acoustics of the environment. A hybrid simulation model is applied combining an image source model and a ray tracing algorithm to generate a binaural room impulse response, which is then convolved with an anechoic sound file (e.g., of a musical instrument). A convenient user interface is introduced by integrating the simulation engine into the 3D modeling program SketchUp. This makes it possible to easily position virtual sound sources and select various source directivities inside a modifiable room model. The listener receives immediate perceptual feedback and is able to examine the effects of varying virtual sound source locations and source directivities in different room acoustic environments.

Contributed Paper

9:40

3aAAa6. Recent advancements in massive multi-channel auralization. Jens Ahrens and Hagen Wierstorf (Univ. of Technol. Berlin, Ernst-Reuter-Platz 7/TEL 18, Berlin 10587, Germany, ahrjens@gmail.com)

Massive multi-channel auralization approaches like Wave Field Synthesis and Higher Order Ambisonics experienced a pronounced hype in the late 2000s during which the primary research goal was maximizing the physical accuracy of the synthetic sound fields that they create. The hype eventually faded as the achievable advancements turned out to be limited due to fundamental restrictions. Though, activities are still being pursued in the domain with the focus shifted towards perception of synthetic sound fields. This talk gives an overview over current activities, which aim at understanding localization, timbre, and spatial impression in general. The results show that localization performance in synthetic sound fields is close to the performance in real sound fields. Timbre and spatial impression exhibit impairments that are directly linked to the physical limitations of the employed systems. Promising options for improvements regarding the synthesis of artificial reverberation are discussed.

Session 3aAAb

Architectural Acoustics: Worship Space Acoustics: Three Decades of Design

David T. Bradley, Cochair
Physics + Astronomy, Vassar College, 124 Raymond Avenue, #745, Poughkeepsie, NY 12604

Erica E. Ryherd, Cochair
University of Nebraska, 1110 S 67th St., Architectural Engineering, Omaha, NE 68182-0816

Lauren M. Ronsse, Cochair
Audio Arts and Acoustics, Columbia College Chicago, 33 E. Congress Pkwy, Suite 601, Chicago, IL 60605

Chair's Introduction—9:35

Invited Papers

9:40

3aAAb1. The soundscape of worship. Gary W. Siebein and Kara A. Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

Soundscape theory provides insights into the architectural acoustic design of religious buildings that demonstrate how acoustical issues can be integrated within worship spaces of the 21st century. Eliade (1959) conceives of worship spaces as a sacred enclosure where communication with the God is made. The spaces must have a door to the world above, by which the God can descend to earth and man can symbolically ascend to heaven. Sometimes this is manifest as one confronts his or her God alone, in silence; sometimes, it is manifest in the joyous singing and praying or sad wailing of the entire community joined in corporate celebration or solemnity during momentous events. If one pays careful attention to the group involved with the design of a facility, they can tailor the architectural, acoustical, and electronic systems design to provide a unique acoustical identity that expresses the individual aspirations of the particular church. The technical tools for measurement and assessment of existing conditions, modeling of proposed changes, and simulating the anticipated results so evaluations can be made while the design is still in a computer are readily available to allow consultants to develop acoustical aspects of worship spaces that are unique for each project.

10:00

3aAAb2. How classical ideas inform 21st century architectural acoustics. Timothy Foulkes (Cavanaugh Tocci Assoc, Sudbury, MA) and John Prokos (Gund Partnership, 47 Thorndike St., Cambridge, MA 02141, JohnP@gundpartnership.com)

This presentation was inspired by John Prokos' essay in *Worship Space Acoustics: 3 Decades of Design*. It will be a joint presentation by Prokos, an architect and Foulkes, an acoustician. It was Marcus Vitruvius Pollio, the great First Century Roman architect and engineer, who first set forth the famous triumvirate of *firmness* (structure), *commodity* (usefulness), and *delight* (beauty) as the ideals of the architect. Even in today's digital age, these three tenets are as applicable as ever, nowhere more so than in architectural acoustics. This session will consider examples of buildings and spaces whose acoustics are as compelling as their visual design. For *firmness*, we will look at several examples including Gund Partnership's athletic center at Kenyon College, whose dramatic vaulted glass and steel structure nonetheless allow excellent acoustics. For *commodity*, among the projects we present is the Young Israel Symagogue in Brookline, MA, a collaboration between the two presenters, which has the flexibility to accommodate varied worship services and gatherings to meet the congregations's changing needs. Among the buildings demonstrating *delight* is Cavanaugh Tocci's work on the 1100 seat Oratory at Ave Maria University in Florida, a classically inspired church that seeks to replicate the acoustics of the great European cathedrals.

10:20

3aAAb3. Balancing sight and sound in heritage worship buildings. Dan Clayton (Clayton Acoust. Group, 2 Wykagyl Rd., Carmel, NY 10512-6224, mail@claytonacoustics.com)

Buildings for traditional religious gathering and time-honored worship practice require a balance of highly valued acoustical qualities such as reverberance for liturgical music, ensemble for choral singing, responsiveness for congregational participation, and clarity for intelligible speech. Geometry, dimensions, proportions, cubic volume, and boundary materials are critical elements of acoustical success. Many old European worship buildings are admired for their particularly fine blending of these criteria, and often cited as benchmarks for how buildings in the United States should look and sound. Although many heritage U.S. worship buildings look like their European precedents in terms of layout, shape, and size, a critical design element was modified, upsetting the acoustical balance. Boundary materials became lighter and thinner, making construction easier, faster, and less expensive. Acoustical upgrade of these existing buildings in the context of historic preservation/restoration brings additional design complexity, requiring equal measures of compromise from owner, users, acoustician, architect, and engineers. The acoustician's dilemma becomes "what we see may not be what we hear," as acquired acoustical expectations are upended by actual conditions. This paper will describe these differences and explore approaches to acoustical enhancement within limitations of the buildings themselves plus further constraints of contemporary preservation practice.

10:40

3aAAb4. Worship space acoustics: Design needs, trends, and issues for sacred spaces, relative to traditional and contemporary worship styles. Scott Riedel and Craig Schaefer (Acoust. Consultants, Scott R. Riedel & Assoc., Ltd., 819 N Cass St., Milwaukee, WI 53202, riedel@riedelassociates.com)

Concerns and goals for worship space acoustic design include the need for clear, intelligible speech, well blended, balanced, and projected music, and encouraging robust participation by the congregation in sung and spoken parts of a service. Also important are the attenuation of noise and control of acoustical anomalies that could interrupt, mask, or distort the critical expressions of speech and music. Achieving these goals includes careful attention to such factors as various cubic air volumes, unique geometric forms, functional proximities, and interior finish material selections. These factors are important to both improving existing rooms and in creating new designs. A current challenge is that of accommodating different musical styles (from traditional compositions and instruments to contemporary and ethnic forms) within the same space. Practical issues may include client understanding of differences between A/V systems and their capabilities vs. architectural room acoustics, and realities of budget limitations and value engineering. Before and after case-study examples exploring these goals, issues, and design solutions will be presented and discussed, including examples from the new publication, "Worship Space Acoustics, Three Decades of Design."

11:00

3aAAb5. Practical lessons in acoustical design learned from recent worship space projects. Benjamin Markham (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

In the last 25 years of worship space acoustics consulting projects (and building on experiences of the 25 years before that), consultants at Acentech have learned many important lessons about these unique and important spaces. Throughout that time period, the evolution of worship style has necessitated thoughtful new design approaches to worship space acoustics. Acentech consultants have learned lessons regarding the complex interplay between support for congregational singing and appropriate reverberation control for contemporary worship music. The equally complex interplay between traditional music and intelligibility of a preacher's speech has also taught profound and lasting lessons. Silence plays a critical and fundamental role in promoting a contemplative and prayerful environment—another important lesson learned. Using examples from the Acoustical Society's new Worship Spaces book, this presentation will illustrate practical lessons from these and other projects in Acentech's recent experience with worship facilities.

11:20

3aAAb6. Worship space acoustics: Three decades of design. Walid Tikriti (Acoustonica, LLC, 33 Pond Ave., Ste. 201, Brookline, MA 02445, wtikriti@acoustonica.com)

The presentation discusses the acoustics and noise control design for new construction and existing worship space renovation and the challenges during the design stage. The first project is an addition of a chapel to an existing church. Treating the hard surfaces on the walls and ceiling proved challenging due to the historic nature of the building. The second project is an addition of a 1800 sq. ft. multipurpose room with a full glass wall. The Glory Be Hall space is intended to provide an acoustically pleasant space for various programs. The third project is a new construction of a 550 seat worship space. The acoustics design focused on achieving a balance between the amplified sound system, the background noise level from the mechanical equipment, the noise isolation from adjacent spaces including nursery and public spaces.

11:40

3aAAb7. Design and renovation of worship spaces at Fourth Presbyterian Church in Chicago. Dawn Schuette and Jennifer Nelson Smid (Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, jnelson@thresholdacoustics.com)

Fourth Presbyterian Church is among Chicago's most iconic buildings. Designed by Ralph Adams Cram, it was dedicated in 1914. Threshold Acoustics worked alongside Gensler on a 5-story addition to the Fourth Church facilities that provided needed worship and program space in celebration of their centennial. The Genevieve and Wayne Gratz Center was included as a case study for the publication "Worship Spaces Acoustics: Three Decades of Design." It houses the 350-seat Buchanan Chapel, which is used as overflow for holiday services, intimate worship services, Children's Chapel, music performances, and weddings. Threshold Acoustics is currently working with Fourth Church on the restoration of their historic Skinner Organ in the sanctuary. An acoustic study of the sanctuary was completed to evaluate and determine the current challenges the organ faces to carry sound to all locations within the Nave. The renovations to the organ and chamber are underway and are to be finished by December 2015 for holiday services. This paper will describe Threshold's acoustic design for the Gratz Center and Sanctuary.

Session 3aAO

Acoustical Oceanography: Munk Award Lecture

Andone C. Lavery, Chair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 98 Water Street, MS 11, Bigelow 211, Woods Hole, MA 02536

Chair's Introduction—10:55

Invited Paper

11:00

3aAO1. Ocean acoustic tomography: Past, present, and maybe future. Carl Wunsch (Earth and Planetary Sci., Harvard Univ., Cambridge, MA 02138, carl.wunsch@gmail.com)

This talk will describe some of the origins of what became known as ocean acoustic tomography, and how it has evolved in the intervening years. Some published mis-conceptions about its beginnings are somewhat interesting. The possibilities of its having a more central role in the determining the evolution of the ocean in future climate raise fascinating technical problems that probably have equally fascinating solutions.

WEDNESDAY MORNING, 4 NOVEMBER 2015 CLEARWATER, 8:00 A.M. TO 11:05 A.M.

Session 3aBAa

Biomedical Acoustics: Sonothrombolysis

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

Kenneth B. Bader, Cochair
Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3935, Cincinnati, OH 45267-0586

Chair's Introduction—8:00

Invited Papers

8:05

3aBAa1. Non-invasive thrombolysis using histotripsy beyond the "intrinsic" threshold (microtripsy). Xi Zhang (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., Rm. 1107 Gerstacker Bldg., Ann Arbor, MI 48109), Hitinder Gurm (Interventional Cardiology, Univ. of Michigan, Ann Arbor, MI), Gabe Owens (Pediatric Cardiology, Univ. of Michigan, Ann Arbor, MI), and Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI 48109, zhenx@umich.edu)

Histotripsy has been demonstrated as a non-invasive, drug-free, image-guided thrombolysis method using cavitation alone. Microtripsy is a new histotripsy approach, where cavitation is generated using 1-cycle ultrasound pulses with negative pressure exceeding a threshold intrinsic to the medium. We hypothesize that, using microtripsy, cavitation can be generated entirely within the vessel lumen, without contacting the vessel wall. Microtripsy was used to create a flow channel through a clot in a vessel-mimicking phantom by scanning the therapy focus through the clot at a pre-set interval. Different scan intervals, doses, and strategies (single focus vs. electrical-steered multi-foci) were tested in both unretracted and retracted clots. The flow channel size, thrombolysis rate, and clot debris particle sizes were measured. A cavitation cloud and flow channel was successfully generated and completely confined within the 6.5 mm-diameter vessel lumen through unretracted and retracted clots. A 1-4 mm channel was created in unretracted clots, at thrombolysis rate of 3.3 min/cm. Using an electrical-steered multi-foci method, a 1–2.5 mm channel was generated through retracted clots, at thrombolysis rate of 5.5 min/cm. The debris particles generated were no greater than 150 μm. The results show the potential of microtripsy for precise and effective clot recanalization, minimizing risks of vessel damage and embolism.

8:25

3aBAa2. Insights into mechanisms of sonothrombolysis using high speed imaging. Flordeliza Villanueva (HVI, UPMC, 200 Lothrop St. PUH A351, Pittsburgh, PA 15213, villanuevafs@upmc.edu)

Thrombotic arterial occlusion is the major cause of acute cardiovascular syndromes such as stroke and myocardial infarction. Exposing the thrombus to microbubbles in the presence of ultrasound facilitates thrombus disruption, and is thus a potentially powerful therapeutic strategy for thromboembolic diseases. However, optimization of "sonothrombolysis" is constrained by a lack of understanding of underlying mechanisms for blood clot disruption in response to ultrasound-induced microbubble vibrations. We tested the hypothesis that inertial cavitation induces mechanical clot disruption by optically characterizing lipid microbubble interactions with thrombus + ultrasound using an ultra-high speed microscopy imaging system capable of imaging at MHz frame rates. A microscope/acoustic stage was used to hold an experimentally created thrombus and microbubbles, which were insonified (1 MHz) during synchronized high speed imaging. Large amplitude microbubble oscillations in response to an inertial cavitation regime caused thrombus deformation and pitting. These data implicate a direct mechanical effect of oscillating microbubbles on mediating clot disruption.

8:45

3aBAa3. Combined lysis of thrombus with ultrasound and systemic tissue plasminogen activator for emergent revascularization in acute ischemic stroke (CLOTBUST-ER): An update. Andrei V. Alexandrov (Neurology, UTHSC, 855 Monroe Ave., Ste. 415, Memphis, TN 38613, avalexandrov@att.net)

Background: Continuous exposure of intracranial arterial occlusions to pulsed wave ultrasound enhances tissue plasminogen induced recanalization. Our hypothesis is that sonothrombolysis can improve functional outcomes of stroke patients receiving tPA therapy. Methods: The primary objective is to evaluate the efficacy of a novel transcranial ultrasound device and systemic tPA (Target) compared to systemic tPA alone (Controls) in subjects with acute ischemic stroke and NIHSS scores 10 or greater. This is a randomized (1:1), placebo-controlled, multi-site, phase 3 clinical trial to evaluate the efficacy and safety of a novel ultrasound device, as an adjunctive therapy to tPA treatment in subjects with acute ischemic stroke: total projected enrollment 824 patients with interim analyses at 1/3 and 2/3 of enrollment. Current Status: CLOTBUST-ER had active enrollment at approximately 70 sites in 14 countries worldwide. DSMB recommended to stop the trial after second interim analysis. Functional outcomes are still being assessed via modified Rankin Scores at 90 days and the primary endpoint will be analyzed using ordinal shift statistical analysis. Final results will be presented at ISC 2016. Conclusions: CLOTBUST-ER is the first phase three multinational randomized blinded clinical trial of sonothrombolysis for treatment of acute ischemic stroke. ClinicalTrials.gov Trial Registry ID: NCT01098981.

9:05

3aBAa4. Microbubble pumps: Ultrasound theragnostic agents. Christy K. Holland, Himanshu Shekhar, and Kenneth B. Bader (Internal Medicine, Div. of Cardiovascular Health and Disease, and Biomedical Eng. Program, Univ. of Cincinnati, Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

Cardiovascular disease is the number one cause of death worldwide, and thrombo-occlusive disease is a leading cause of morbidity and mortality. Ultrasound has been developed as a tool to induce the release, delivery, and enhanced efficacy of a thrombolytic drug (rt-PA) and bioactive gases from echogenic liposomes. By encapsulating drugs into micron-sized and nano-sized liposomes, the therapeutic can be shielded from degradation within the vasculature until delivery is triggered by ultrasound exposure. Insonification accelerates clot breakdown in combination with rt-PA and ultrasound contrast agents, which nucleate sustained bubble activity, or cavitation. Mechanisms for ultrasound enhancement of thrombolysis, with a special emphasis on cavitation and radiation force, will be reviewed. The delivery of bioactive gases from echogenic liposomes to promote vasodilation and cytoprotection will also be discussed.

9:25

3aBAa5. Microscale interactions between ultrasound stimulated microbubbles and the fibrin networks of clots. David Goertz, Christopher Acconcia (Univ. of Toronto, S665a, 2075 Bayview Ave., Toronto, ON m4k1x5, Canada, goertz@sri.utoronto.ca), and Ben Leung (Sunnybrook Res. Inst., Toronto, ON, Canada)

While the concept of microbubble mediated sonothrombolysis is now well established, a detailed mechanistic understanding of this process remains both elusive as well as necessary in order to facilitate the development of improved exposure methods. As lytic effects may ultimately arise from microscale bubble-clot interactions, we have employed high-speed microscopy and two-photon microscopy to examine these interactions first in (transparent, fluorescently tagged) fibrin clots, and then in blood clots. Bubble "population" studies in fibrin clots show the prominent role of primary and secondary radiation forces: bubbles are first directed toward the clot boundary, where their concentrations increase and interactions such as clustering and coalescence occur frequently. A subset of bubbles penetrate into the clots, disrupting the fibrin network structure along their paths. Once initiated, the resulting tunnels act in subsequent exposures as conduits for bubbles to enter and access deeper points within the clot. Using optical tweezers, individual bubble experiments reveal that bubble entry into the clots, along with the accompanying network damage and fluid uptake, are a function of the network pore size, bubble size, and the exposure scheme. With blood clots, the erosion surface evolves in a complex manner, involving the ejection of erythrocytes and the development and progression of a cell depleted fibrin network zone. The characteristics of the erosion front are highly dependent on exposure conditions.

9:45

3aBAa6. Molecular mechanisms of the effect of ultrasound on the fibrinolysis of clots. Irina N. Chernysh (Dept. of Cell & Developmental Biology, Univ. of Pennsylvania, 1154 BRB, 421 Curie Blvd., Philadelphia, PA 19104-6058), E C. Everbach (Eng. Dept., Swarthmore College, Swarthmore, PA), Prashant K. Purohit (Dept. of Mech. Eng. and Appl. Mech., Univ. of Pennsylvania, Philadelphia, PA), and John W. Weisel (Dept. of Cell & Developmental Biology, Univ. of Pennsylvania, Philadelphia, PA 19104-6058, weisel@mail.med.upenn.edu)

Our objective was to identify mechanisms for the ultrasound-enhanced acceleration of tissue-type plasminogen activator (t-PA)-induced fibrinolysis of clots. Measurements of turbidity as a function of time, used to characterize quantitatively the effects of ultrasound, showed that the ultrasound pulse-repetition frequency affected clot lysis times, but there were no thermal effects. Ultrasound in the absence of t-PA produced a slight but consistent decrease in turbidity, suggesting a decrease in fibrin diameter due solely to the action of ultrasound, likely caused by an increase in protofibril tension because of vibration from ultrasound. Changes in fibrin network structure during lysis with ultrasound were visualized in real time by deconvolution microscopy, revealing that the network becomes unstable when 30–40% of the network was digested, whereas without ultrasound, the fibrin network was digested gradually and retained structural integrity. Fluorescence recovery after photobleaching during lysis was used to characterize the kinetics of binding/unbinding and transport. Ultrasound causes a decrease in the diameter of fibers due to tension as a result of vibration, leading to increased binding sites for plasmin(ogen)/t-PA. The positive feedback of this structural change together with increased mixing/transport of t-PA/plasmin(-ogen) is likely to account for the observed enhancement of fibrinolysis by ultrasound.

10:05–10:20 Break

Contributed Papers

10:20

3aBAa7. Histotripsy for liquefaction of large extravascular hematomas for fine-needle aspiration: Feasibility study. Tatiana D. Khokhlova, Thomas Matula, Yasser Haider, and Wayne Monsky (Univ. of Washington, 325 9th Ave., Harborview Medical Ctr., Box 359634, Seattle, WA 98104, tdk7@uw.edu)

Intra- and extra- muscular hematomas result from repetitive injury, or sharp and blunt limb trauma. There are currently no short-term treatment options for large hematomas, only lengthy conservative treatment. The goal of this work was to evaluate the feasibility of histotripsy to liquefy extravascular hematomas for subsequent fine-needle aspiration. Cavitation histotripsy (CH) and boiling histotripsy (BH) were applied to liquefy 50-mL clots in an *in vitro* gel phantom. 1 MHz and 1.5 MHz transducers were used with duty cycles of 1–2% for BH and 0.3–0.5% for CH, pulse durations 10 ms for BH, and 3.3–5 çs for CH, treatment duration 5–60 s and peak focal pressures of $p^{+} = 70–100$ *MPa*, and $p^{-} = 15–25$ *MPa*. The liquefied lysate was aspirated with a 18–21 gauge needle. The lysate was analyzed by histology and sized in a Coulter counter. Under the same exposure duration, BH lesions were 1.5–2 times larger than CH lesions. 99% of lysate particulates were smaller than 10 çm. CH-aided liquefaction was slower, but the voids were more regularly shaped, facilitating easier aspiration. A combination of BH and CH may be most optimal for liquefaction. [Work supported by NIH K01 EB 015745 and Washington State Life Science Discovery Fund (Grant No. 3292512).]

10:35

3aBAa8. Fibrin-targeted echogenic liposomes for localized ablation of thrombi with histotripsy pulses. Kenneth B. Bader, Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3935, Cincinnati, OH 45267-0586, Kenneth.Bader@uc.edu), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Tao Peng, David D. McPherson (Dept. of Internal Medicine, Univ. of Texas Health Sci. Ctr., Houston, TX), and Christy K. Holland (Internal Medicine, Univ. of Cincinnti, Cincinnati, OH)

Deep vein thrombosis is a debilitating condition that can result in pulmonary embolism, or post-thrombotic syndrome. Clot lysis can be initiated with histotripsy, a novel form of therapeutic ultrasound that uses the mechanical action of microbubble clouds to ablate tissue. Thrombolytic-loaded echogenic liposomes (t-ELIP) have fibrin targeting capabilities and also entrain gas microbubbles that act as cavitation nuclei. We hypothesize that t-ELIP can nucleate microbubble cloud formation near the clot, providing targeted ablation. Highly retracted porcine whole blood clots were exposed to either histotripsy pulses alone, or histotripsy pulses and t-ELIP between peak negative pressures of 7.5 MPa and 20 MPa. Thrombolytic efficacy was assessed via clot mass before and after treatment. Microbubble cloud activity was monitored with passive cavitation imaging (PCI) during histotripsy exposure. For a given pressure, no significant differences were observed in the thrombolytic efficacy with or without t-ELIP. The dimensions of the bubble cloud, assessed on the PCIs, were significantly reduced in the presence of t-ELIP ($p < 0.05$). Thus, the presence of t-ELIP confines the business end of the bubble activity to the clot. Overall, the combination of histotripsy and t-ELIP is a promising mechanism for targeted ablation of thrombi resistant to thrombolytic drugs.

10:50

3aBAa9. Ultrasound-enhanced thrombolysis: Mechanistic observations. Marie de Saint Victor, Dario Carugo, Constantin Coussios, and Eleanor P. Stride (Univ. of Oxford, IBME, Old Rd. Campus, Oxford, United Kingdom, marie.desaintvictor@pmb.ox.ac.uk)

Ultrasound and microbubbles have been widely demonstrated to accelerate the breakdown of blood clots, but the mechanisms of treatment require further investigation. In particular, there is a need to clarify the effect on the fibrin matrix—the insoluble polymer mesh that determines a clot’s integrity and mechanical properties. The objective of this *in vitro* study was to observe in real-time the mechanisms of microbubble-enhanced sonothrombolysis at the microscale. Fluorescently labeled porcine plasma clots were prepared on a glass coverslip and exposed to different types of microbubbles with or without the fibrinolytic agent recombinant tissue plasminogen activator. A 1 mm thick piezoelectric element was coupled with the glass substrate and driven at the resonant frequency of the system (1.9 MHz), with a duty cycle of 5% and a 0.1 Hz pulse repetition frequency. The acoustic field within the clot was characterized using a fiber optic hydrophone. Changes in the fiber network were monitored for 30 min by confocal microscopy.

Session 3aBAb

Biomedical Acoustics: Therapeutic Ultrasound, Microbubbles, and Bioeffects I

Michael Bailey, Chair
Center for Industrial and Medical Ultrasound, Applied Physics Lab, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Contributed Papers

11:15

3aBAb1. High throughput acoustic and optical characterization of microbubbles for optimized contrast ultrasound imaging. Paul Rademeyer and Eleanor Stride (IBME, Oxford Univ., Old Rd. Campus, IBME, Oxford OX3 7DQ, United Kingdom, paul.rademeyer@eng.ox.ac.uk)

Echogenic particles, such as microbubbles and volatile liquid micro-/nanodroplets, have shown considerable potential in a variety of clinical diagnostic and therapeutic applications. The accurate prediction of their response to ultrasound excitation is however extremely challenging, and this has hindered the optimization of techniques such as quantitative ultrasound imaging and targeted drug delivery. Existing characterization techniques, such as ultra-high speed microscopy, provide important insights, but suffer from a number of limitations; most significantly difficulty in obtaining large data sets suitable for statistical analysis and the need to physically constrain the particles, thereby altering their dynamics. Here, a microfluidic system is presented that overcomes these challenges to enable the measurement of single echogenic particle response to ultrasound excitation with a throughput of 20 samples/second and an uncertainty below 7% in the measurements. Demonstration of optimized contrast ultrasound imaging is shown based on the characterization of over 100,000 individual SonoVue® bubbles.

11:30

3aBAb2. Design, characterization, and performance of a dual aperture, focused ultrasound system for microbubble-mediated, non-thermal ablation in rat brain. Jonathan Sutton, Yui Power, Yongzhi Zhang, Natalia Vykhodtseva, and Nathan McDannold (Dept. of Radiology, Brigham and Women's Hospital, 221 Longwood Ave., Boston, MA 02115, jtsutt@bwh.harvard.edu)

Submegahertz ultrasound (US) with microbubbles (μB) can ablate brain tissue: a low amplitude (< 1 MPa) alternative to thermal ablation. Single element transducers at transcranial frequencies have broad axial profiles compared to the size of targets in small animals. Thus, we sought a system to ablate millimeter volumes in normal and tumorous rat brain non-thermally using US and μB. The system consisted of two transducers oriented at 120°, driven at different frequencies (F = 837 kHz, ΔF = 30 Hz) to reduce the depth-of-field by 78%. To monitor cavitation, a passive detector (650 kHz) was confocally aligned with the therapy field. Targets were registered stereotactically. μB injections (100, 200, and 400 μl/kg) with 5-minute sonications proceeded at acoustic pressures relative to the *in vivo*cavitation threshold (0.3–0.6 MPa) determined *a priori*. Following MRI and sacrifice (1 h, 4 days, 10 days), tissue was fixed and stained. At 1 h, small lesions (<2 mm) were selectively comprised of stenotic capillaries with macrophage infiltration and neuronal damage. As lesions grew, stenotic capillaries within this escalating margin increased, along with neuronal damage. T2* lesions larger than 4 mm exhibited necrosis, cyst, and tissue removal histologically after four days. Persistent, strong (>5 dB) inharmonic energy indicated formation of large ablation volumes along the US path. Low level (<2 dB) or sporadic (<50%) cavitation indicated incomplete ablation. Results and mechanisms, especially with respect to ischemic stroke, will be discussed.

11:45

3aBAb3. An empirical model of size-isolated ultrasound-triggered phase shift emulsions. Karla P. Mercado, Lindsay Snider, Kirthi Radhakrishnan, and Kevin J. Haworth (Univ. of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209, kevin.haworth@uc.edu)

High-speed mechanical agitation is commonly used to produce microbubbles and droplets for ultrasound imaging and therapy. This technique results in a high concentration (~10^{10} particles/mL) of polydisperse particles (less than 400 nm to greater than 15 μm in diameter). Differential centrifugation has been used to isolate microbubbles and droplets of specific sizes. In our prior work, we have isolated droplets between 2 μm and 5 μm. In the current work, we have isolated different sizes of droplets by adjusting centrifugation speeds. Our size-isolation protocol increased the fraction of droplets between 1 μm and 3 μm from 8% for non-centrifuged droplets to 87% for differentially centrifuged droplets. An empirical model for the size distribution after differential centrifugation was developed. The measured fraction of droplets in the supernatant and pellet for all sizes after a single centrifugation was used in the empirical model. There was a 3% difference in the volume-weighted mean diameter of the experimentally measured and empirically modeled size distributions. The coefficient of variations of the experimentally measured and empirically modeled size distributions were 24% and 22%, respectively. The empirical model allows for determining appropriate centrifugation parameters to obtain desired size distributions. [Work supported in part by NIH grant KL2 TR000078.]

Session 3aEA

Engineering Acoustics: Test Facilities and Acoustic Calibration

Roger M. Logan, Chair
Teledyne, 12338 Westella, Houston, TX 77077

Chair's Introduction—8:00

Invited Papers

8:05

3aEA1. ANSI/ASA S1.20-2012, "Procedures for Calibration of Underwater Electroacoustic Transducers," Revision Details. Robert Drake (NUWC, P.O. Box 5029, Newport, RI 02841, orl_preds@yahoo.com)

The ANSI/ASA S1.20-2012 standard, "Procedures for Calibration of Underwater Electroacoustic Transducers," was released in February 2012. It is a revision of American National Standard S1.20-1988 (R2003). An overview of the content of this standard inclusive of the typical primary and secondary open water procedures for determining the measurable characteristics of free-field receive voltage sensitivity and transmitting responses are examined along with highlights of materials that are new to the 2012 revision. In this later category, this presentation provides overview information contained in the S1.20-2012 related to correction factor application (with an emphasis on extension cable usage) and introductory material on measurement uncertainty analysis (along with the identification of common error sources). A cursory look at the informational annexes will also be addressed with an emphasis on the 2012 revision-specific ones related to medium correction factors, nonlinear effects including cavitation, and a standard-target method for calibrating active sonars.

8:25

3aEA2. Underwater acoustic calibration facilities at Applied Research Laboratories, The University of Texas at Austin. Richard D. Lenhart (Appl. Res. Labs., The Univ. of Texas at Austin, PO Box 8029, Austin, TX 78713-8029, lenhart@arlut.utexas.edu) and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The calibration of various underwater sources and receivers is routinely undertaken at The University of Texas at Austin's Applied Research Laboratories using an open water facility, the Lake Travis Test Station, and both indoor and outdoor test tanks on the ARL main campus. Calibrations are also conducted using standing wave facilities and transducer electrical input impedance measurements can be obtained. These facilities and procedures will be described.

8:45

3aEA3. Standard-target calibration of sonars for imaging and scattering measurement: An ISO-standard development. Kenneth G. Foote (Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, kfoote@whoi.edu) and Christian de Moustier (10dBx LLC, San Diego, CA)

Given a standard target of known acoustic scattering properties, the combined transmit–receive transfer characteristics of an active sonar system can be determined by measuring the response due to the target placed at specified positions in the sonar transmit and receive beams. These measurements are repeatable to ± 0.1 dB, establishing an absolute reference when measuring the scattering properties of unknown targets. This standard-target sonar-calibration method is in worldwide use with scientific echo sounders and multibeam sonars that provide the water-column signal, yet no international metrological standard defines it. To prescribe protocols for this method, which has a demonstrated applicability over the frequency range from 1 kHz to several megahertz, an effort was initiated in 2014 within the International Organization for Standardization (ISO) Technical Committee 43 (Acoustics) Subcommittee 3 (Underwater Acoustics). The associated Working Group 4 (Standard-target method of calibrating active sonars) has formulated an outline of the intended ISO standard. The elements of this will be reviewed.

9:05

3aEA4. Acoustic calibration in mass production. Roger M. Logan (Teledyne, 12338 Westella, Houston, TX 77077, rogermlogan@sbcglobal.net)

While calibrations in high volume settings are bound by the same laws of nature that govern laboratory grade measurements, the laws of economics bring their own set of challenges to the bench. This presentation will explore the art and science of making efficient acoustic measurements in small air-filled tanks using mostly COTS equipment.

9:25

3aEA5. Investigation and correction for the calibration error of using two-microphone impedance tube method on low absorption materials. Yi Zhang, Tongan Wang, and Kristopher Lynch (Acoust., Vib. & Community Noise, Gulfstream Aerosp., 500 Gulfstream Rd., Savannah, GA 31408, Yi.Zhang1@gulfstream.com)

The two-microphone impedance tube method has been widely used for determining the sound absorption coefficients of materials. In this method, a calibration procedure is performed to correct the amplitude and phase mismatch of the two microphones, through repeated measurement of a specimen with the two channels interchanged. Both ASTM E1050-12 and ISO 10534-2 suggest the use of a highly absorptive material as the calibration specimen and the calibration, once complete, is valid for all successive measurements. When following the Standards, however, test materials of low absorption capability exhibited a pattern of peaks and dips in the absorption coefficient curve at certain frequencies, which are obviously not a part of the material's own behavior. This paper will discuss the cause of these peaks and dips and also suggest a simple solution to the issue. Using the new method, one is able to obtain smooth absorption coefficient curves for low absorption test materials just as expected.

9:40

3aEA6. Calibration of high frequency MEMS microphones and pressure sensors in the range 10 kHz–1 MHz. Sébastien Ollivier (LMFA - UMR CNR 5509, Univ. Lyon 1, Ctr. Acoustique - Ecole Centrale de Lyon, 36 Ave. Guy de Collongue, Ecully 69134, France, sebastien.ollivier@univ-lyon1.fr), Petr V. Yuldashev (Faculty of Phys., M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Cyril Desjouy (LMFA - UMR CNRS 5509, Ecole Centrale de Lyon, Ecully, France), Maria Karzova, Edouard Salze (LMFA - UMR CNRS 5509, Ecole Centrale de Lyon, Lyon, France), Alexandra Koumela, Libor Rufer (TIMA Lab. (CNRS, G-INP, UJF), Grenoble, France), and Philippe Blanc-Benon (LMFA - UMR CNRS 5509, Ecole Centrale de Lyon, Ecully, France)

In the context of both nonlinear acoustics, and downscaled acoustic or aero-acoustic experiments, the characterization of the high frequency response of microphones and pressure sensors remains a critical challenge. In the case of the design of new MEMS microphones and shock pressure sensors with response in the frequency range of 10 kHz–1 MHz, this question was addressed by the definition of a new calibration method based on a spark source that generates spherical weak shock acoustic pulse. Waves are short duration non-symmetric N-waves with duration of about 40 microseconds and front shock rise time of the order of 0.1 microsecond. Taking advantage of recent works on the characterization of such pressure waves using an optical interferometer, and considering non linear propagation of weak shockwaves, we were able to estimate the incident pressure wave in the range of 10 kHz–1MHz. Hence, from the output voltage of the microphones, the frequency response was obtained in this range. The method applies whatever the transduction principle and the sensor mounting. [Work supported by the French National Agency for Research (SIMI 9, ANR 2010 BLANC 0905 03, and LabEx CeLyA ANR-10-LABX-60/ANR-11-IDEX-0007).]

9:55–10:10 Break

10:10–10:30 Panel Discussion

WEDNESDAY MORNING, 4 NOVEMBER 2015 GRAND BALLROOM FOYER, 10:00 A.M. TO 12:00 NOON

Session 3aED

Education in Acoustics: Undergraduate Research Exposition (Poster Session)

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

Joseph F. Vignola, Cochair
Mechanical Engineering, The Catholic University of America, 620 Michigan Ave., NE, Washington, DC 20064

Posters will be on display and all authors will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

3aED1. Experimental investigation of nonlinear wave behavior in a tensegrity mast. Joy Westland, Andrew A. Piacsek, and Peter Zencak (Dept. of Phys., Central Washington Univ., Ellensburg, WA 98926-7422, westlandj@cwu.edu)

First described by Fuller and Snelson in the mid-twentieth century, tensegrity structures comprise a network of elements that alternately support compression (struts) and tension (cables). When the cable elements are prestressed, tensegrity structures maintain a stable form, regardless of orientation. If external forces are applied, the structure will deform both locally and globally, with the stress distributed throughout the structure. Removal of the external forces will result in the structure regaining its original shape. However, the deformation is not a result of elastic strains in the individual elements, but rather a reorientation of the elements, and the restoring force is nonlinear. To characterize the effect of this nonlinearity on dynamic behavior, a tensegrity mast was constructed consisting of 60 aluminum bars, each of length 11 cm; the total mast length is approximately one meter. A mechanical shaker mounted at one end is used to drive the mast into resonance, as well as to create longitudinal pulses. Frequency response measurements are consistent with a stiff nonlinearity and evidence is shown for harmonic generation during pulse propagation. The effect of pre-stress tension is also investigated and discussed.

3aED2. Electroacoustical evaluation of a prototype headphone amplifier for educational purposes. Lúcia C. da Silva (Centro de Tecnologia, Universidade Federal de Santa Maria, Rua Lauro Bulcão, 1715, São Sepé, RS 97340-000, Brazil, lucodasil@hotmail.com.br), Ricardo Brum, Moisés dos Santos Canabarro (Centro de Tecnologia, Universidade Federal de Santa Maria, Santa Maria, RS, Brazil), and Stephan Paul (Centro de Engenharia da Mobilidade, Universidade Federal de Santa Catarina, Florianópolis, SC, Brazil)

In several classroom activities, it is required to feed a signal simultaneously to several headsets to have students listening to the same sound signal, e.g., to show binaural effects. To drive several headphones, an appropriate headphone amplifier is needed. The aim of this work was to build a prototype of a simple headphone amplifier based on a publicly available project around different types of operational amplifiers and test its electrical and acoustical performance when used with typical headphones. Frequency response function, total harmonic distortion, and signal-to-noise ratio were evaluated. Among the different OPAMPs the OPA2134 showed the best performance, especially because of the very flat frequency response function in the hearing range, low harmonic distortion, and superior SNR (−83 dBFS). Acoustical test were therefore performed based on the amplifier with the OPA2134 only, using a Sennheiser HD 600 headphone and a Sennheiser KR4 microphone. THD measured with this configuration ranged from −86 dBC@2.5 kHz and global SPL = 70 dB to −35 dBC@100 Hz and global SPL = 110 dB. At all, the test showed that the amplifier circuit is appropriate for the required application, taking into account that it is very simple and can be mounted easily by the students.

3aED3. Acoustic and sediment data in the southern New England Bight. Chitanya Gopu (Dept. of Mech. Eng., Boston Univ., Boston, MA 02215, chitanya@bu.edu), Gopu R. Potty, James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and James F. Lynch (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Wood Hole, MA)

This study provides a review of the acoustic and ocean bottom sediment data collected during the Shelf Break Primer experiment conducted in 1996. The location of the experiment was in the southern New England Continental Shelf called the “New England Mud Patch.” The mud patch is a 13,000 square kilometer area covered by fine-grained sediment. Previous surveys in this area have estimated the thickness of this fine grained sediment deposit to be as much as 13 m. This layer of sediment rests on a reflector that is geomorphically similar to and continuous with the Holocene transgressive sand sheet, which is exposed on the shelf to the west of this area. This fine-grained sediment layer, which is oriented in an east-west direction seaward of the 55–65 m isobath, contains more than 30% silt and clay. During the Primer experiment, broadband acoustic sources were deployed in the western side of the “mud patch” along and across the continental shelf. The acoustic data collected on a vertical line array will be analyzed. Sediment data from gravity cores from the area will also be presented. [Work sponsored by Office of Naval Research, code 322 OA.]

3aED4. Estimation of the effective perceived noise level during approach of Brazilian AMX A-1 military aircraft. Bernardo H. Murta, Gil F. Greco (Universidade Federal de Santa Maria, Rua Dezenove de Novembro, 289, 302, Santa Maria, Rio Grande do Sul 97060160, Brazil, be.murta@gmail.com), Stephan Paul (Universidade Federal de Santa Catarina, Florianopolis, Brazil), Jean C. Bernardi, and Matheus Pereira (Universidade Federal de Santa Maria, Santa Maria, Brazil)

Aircraft noise is an important cause of annoyance mainly in regions close to airports or military airbases. Civil aircraft data of sound emission, such as EPNL, are commonly available to the public; however, for military aircrafts such data are not easily accessible. Noise emission is commonly quantified in terms of Equivalent Perceived Noise Level during approach, low altitude overflight, and simulated take-off. This study presents the results of the estimation of the Equivalent Perceived Noise Level (EPNL) for the AMX-A1 aircraft during approach at Santa Maria’s Airbase, in south Brazil. Since there is an University and its Hospital on the surroundings of the airbase, the approach is the most annoying situation. The measurements were made at the vicinities of the airbase based on the equivalent procedures established by ICAO where calibrated wave signals were recorded and processed, leading to results of EPNL of the order of 134 EPNdB. The calculated levels were found to be coherent for military aircrafts and shall be validated with further investigation.

3aED5. Sonothrombolysis of porcine blood clots using 1 MHz pulsed ultrasound. Atousa Nourmahnad, Luke Barbano, and Erich C. Everbach (Eng. Dept., Swarthmore College, 500 College Ave., Swarthmore, PA 19081, eceverbach@comcast.net)

Blood clots block proper blood flow in vessels and often lead to fatal health conditions such as cardiac ischemia and stroke, especially in an aging population. Because ultrasonic fields can cause cavitation of fluids, administration of ultrasound and microbubble contrast agents can induce intravascular thrombolysis (dissolution of blood clots). Quantitative and qualitative data were collected on how a porcine blood clot dissolves when exposed to Definity™ microbubbles and 1 MHz pulsed ultrasound. In addition, bubble dynamics in response to the measured ultrasonic wave field were modeled using a modified Gilmore equation. The results allowed us to optimize parameters and better understand the interaction of the clot fibrin structure and the movement of microbubbles through it. Advances in the understanding of sonothrombolysis can help transform its clinical application in patients efficiently, especially in instances where life depends on rapid dissolution of a thrombus.

WEDNESDAY MORNING, 4 NOVEMBER 2015 GRAND BALLROOM 1, 8:30 A.M. TO 11:20 A.M.

Session 3aNS

Noise, ASA Committee on Standards, and Psychological and Physiological Acoustics: Role of Fit-Testing Systems for Hearing Protection Devices

Melissa Theis, Cochair
Air Force Research Laboratory, 426 Crusader Drive, Dayton, OH 45449

Elizabeth McKenna, Cochair
Air Force Research Lab., 2610 Seventh St., Wright-Patterson AFB, OH 45433

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—8:30

Invited Papers

8:35

3aNS1. Operational challenges associated with the use of fit test systems in the U.S. Air Force. Melissa Theis (Oak Ridge Inst. for Sci. and Education, 426 Crusader Dr., Dayton, OH 45449, melissa.theis.1.ctr@us.af.mil) and Elizabeth McKenna (The Henry M. Jackson Foundation for the Advancement of Military Medicine, Wright Patterson AFB, OH)

Hearing protection fit test systems provide immediate and direct feedback on the goodness-of-fit for a specific individual based on the personal attenuation rating (PAR) obtained from the measurement. Fit test systems may provide improved retention of hearing protection training, particularly when used in combination with face-to-face training. One challenge for the U.S. Air Force is determining when face-to-face training should occur, and how detailed this training should be to maximize the benefits for end users. A second challenge is determining if various hearing loss patterns render PAR data inaccurate, because many Air Force personnel have hearing loss. The ability of individuals with hearing loss to achieve a desired PAR for a workplace may be influenced by the dynamic range of the system, particularly when measuring the protected condition. This presentation will discuss how the two challenges mentioned above must be investigated before implementing fit test systems as a hearing preservation tool for U.S. Air Force personnel, and discuss some preliminary data obtained in these areas.

8:55

3aNS2. Assessing attitudes toward use of hearing protection devices and effects of an intervention based on results of fit testing. Pegeen Smith (Tech. Service, 3M Co., 620 Holly Rd., Cadillac, MI 49601, psmith4@mmm.com)

As field attenuation estimation systems (FAESs) have become more prevalent, there has been a call from industry to quantify the value of including them into hearing conservation programs. This presentation describes a study that assessed attitudes toward the use of hearing protection devices (HPDs) and the effect of an educational intervention based on results of fit testing by comparing the personal attenuation rating (PAR50) before and after intervention. Employees (n = 327) from a large metal container manufacturer at four geographic locations were tested with a FAES to identify workers (n = 91) needing the intervention. PAR50 values significantly increased

from baseline to post-intervention ($p < 0.001$, 15.1 to 26.9) and at six-month follow-up ($p < 0.001$, 95% CI = −11.2, −6.3). However, perceived self-efficacy (SE) scores in using HPDs significantly declined from baseline to post-intervention ($p = 0.006$, 95% CI = 0.3, 1.9) but were not significantly related to PAR50. Therefore, a FAES can assist the health and safety professional to identify workers who are at high risk (low PAR50), teach the proper fit and use of HPDs, and enhance hearing protector selection.

9:15

3aNS3. Viability of large-scale fit-testing in the U.S. Navy and Marine Corps. Jeremy Federman (SUBMED, NSMRL, Bldg 141 Trout Ave, NAVSUBASE NLON, Groton, CT 06349-5900, jeremy.s.federman.civ@mail.mil)

In support of the Navy and Marine Corps initiatives to prevent military and civilian personnel from suffering hearing loss due to noise exposure, and to ensure hearing readiness and fitness for duty in the military and civilian workforce, this project explored the potential to implement a fit testing procedure for objectively measuring hearing protection devices (HPDs) performance on individuals. Due to the relative uniformity of noise exposure during recruit training, relevant training and experience, and likelihood of participants having hearing within normal limits, 320 training recruits at USMC MCRD Parris Island were included in Phase 1 and 900 in Phase 2. Factors of interest were fit testing pass-fail rates, test duration, and training. The pass-fail criterion attenuation level was set at 25 dB based on the output level of the M4 weapon used during basic training (Amrein & Letowski, 2012; NIOSH, 2014). Both test duration and PAR estimation accuracy were considered in evaluating the practicality of implementing a fit-testing program. Results of objective fit testing were considered as a metric of training effectiveness and HPD functionality.

9:35

3aNS4. Hearing protection fit-test system pilot study: Results and recommendations for the hearing conservation program. Quintin Hecht and Matthew Williams (Public Health Consultation, USAF School of Aerosp. Medicine, 2510 5th St., Wright-Patterson AFB, OH 45433-7913, quintin.hecht.1@us.af.mil)

Since the inception of hearing protection device fit-test systems, long-term training benefits have been scarcely investigated. A hearing protection fit-test system was evaluated for performance and training benefits in a pilot study among U.S. Air Force members randomly sampled from those in the hearing conservation program. Subjects were randomly assigned to control or intervention. Both groups received an initial and a 6-month follow-up appointment. At the initial appointment, the control group received standard training for hearing protection, while the intervention group was fit-tested, trained, and fit-tested again. At the follow-up appointment, both groups received fit-testing. Results of the intervention group's initial appointment will be examined and the two groups' follow-up fit-test results will be compared. These findings will be discussed in relation to effects on training and retention. Methods for conducting this study and a descriptive analysis of the subject population will also be reviewed. Recommendations for fit-testing integration into a hearing conservation program will be provided based on our study results and other literature.

9:55–10:10 Break

10:10

3aNS5. Comparison sound-field measurements of hearing protector attenuation and fit-test systems. William J. Murphy and David C. Byrne (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)

The American National Standards Institute (ANSI), accredited S12 standards committee for noise, working group 11 is developing a standard to characterize the performance of hearing protector fit test systems. An inter-laboratory study was conducted to compare sound field measurements of real-ear attenuation at threshold (REAT) tests of hearing protection with three hearing protection fit-test systems. The draft ANSI standard for personal attenuation ratings from fit-test systems includes adjustments to the fit test system personal attenuation rating for the accuracy and precision of the measurement. As well, adjustments are proposed for the effect of fitting and for spectral variability of effective attenuation. In the inter-laboratory study, two fit-test systems exhibited precision of about 2 dB while the third system had a precision of about 4 dB. The third system underestimated the sound field attenuation by about 2 dB. Because the inter-laboratory study tested two fittings of the hearing protector with each fit-test system, the variability of individual fitting is described as an average of the differences of first and second tests. This paper will also evaluate the effect of the noise spectrum for the three fit-test systems.

10:30

3aNS6. Earplug fit testing: Practical examples. Theresa Y. Schulz (Honeywell Safety, 9218 Brookwater Circle, College Station, TX 77845, Theresa.Schulz@Honeywell.com)

Preventing noise-induced hearing loss at the workplace appears deceptively easy. Well-intentioned professionals often assume that providing hearing protectors and administering audiograms are, by themselves, preventive measures. But occupational hearing loss often continues unabated. This presentation shows data from companies that have used best practice techniques to stop noise-induced hearing loss with fit-testing and individual training as their hearing conservation tools. These tools can change the attitudes and behaviors of earplug users, young and old.

10:50–11:20 Panel Discussion

Session 3aPA

Physical Acoustics: General Topics in Physical Acoustics I

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

Contributed Papers

8:00

3aPA1. Transmission and reception of ultrasound via a polymer film. Hironori Tohmyoh and Shota Mukaimine (Dept. of Nanomechanics, Tohoku Univ., Aoba 6-6-01, Aramaki, Aoba-ku, Sendai, Miyagi 980-8579, Japan, tohmyoh@ism.mech.tohoku.ac.jp)

We report on the changes in the waveform of the echo which was transmitted and received via a polymer film inserted between the water and the solid samples. Generally, the frequency characteristics of an ultrasonic transducer are fixed because the piezoelectric element embedded in the transducer is unchangeable. This study is aimed to enhance the flexibility of ultrasonic transducers by freely changing the frequency components of the echo. For this purpose, we paid attention to the frequency dependence of the echo transmittance among three media. A theoretical model for the ultrasonic transmission system comprising the water, the film and the solid sample was developed and the validity of the model was verified by experiments where the ultrasound excited by the ultrasonic transducer was transmitted into the steel samples via a polymer film. The amplitude of the echo was enhanced by the insertion of the film and its frequency components were modulated toward the higher frequency side. It was experimentally confirmed that the waveform of the echo obtained from the sample via a film could be accurately predicted by the theoretical model developed.

8:15

3aPA2. Uncertainty calculations of pressure sensitivity of one inch microphones using pressure field method at the National Metrology Institute of Egypt. Rabab S. Ahmed, Hany A. Shawky, Rabab S. Youssef, Tarek M. El-Basheer, and Hatem K. Mohamed (Acoust., National Inst. for Standards (NIS), Tersa St., El Harm, Giza, Egypt, Cairo 12211, Egypt, ruby01986@yahoo.com)

Primary calibration method is used by relatively few laboratories such as national calibration laboratories and few large automotive, space, or governmental organizations, which work at high technological level. The National Metrology Institute of Egypt (NIS) has developed a pressure calibration system that used for calibration of pressure sensitivities. In this study, we estimate the uncertainty of unknown one inch condenser microphone using two known references one inch condenser microphones according to international standard. IEC 61094-2009 gives more information's and details on the uncertainty calculations. In this method, a wide frequency range with a high accuracy and repeatability were achieved.

8:30

3aPA3. The research on piezoelectric cylindrical oscillating transducer for dipole acoustic logging. Yinqiu Zhou, Xiuming Wang, and Yuyu Dai (Inst. of Acoust., Chinese Acad. of Sci., 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, zhouyinqiu@mail.ioa.ac.cn)

Acoustic transducer, which is a core component of acoustic logging tool, determines the development of acoustic logging technology. In order to realize the goal of dipole acoustic logging transducer with the features of low frequency, broadband and high radiation power, a cylindrical oscillating dipole acoustic transducer using an actuator of a group of different trilaminar piezoelectric bending bars is designed and studied. Modal and harmonic response analyses of the whole transducer are conducted by using the finite element method, including analyses of input conductance of transducer in air and water, transmitting voltage response and directivity patterns. Besides, the proposed transducer is fabricated and tested to compare with numerical simulation results. Moreover, the numerical results of the proposed transducer are compared with that of traditional dipole transducer. The comparison result indicates the proposed dipole transducer is characterized by low frequency, broadband, and high radiation power, which is feasible in the application of dipole acoustic logging.

8:45

3aPA4. Manipulation and levitation of particles with acoustic vortices. ZhenYu Hong (Dept. of Appl. Phys., Northwestern PolyTech. Univ., Xi'an, China), Asier Marzo, Sri Subramanian (Dept. of Comput. Sci., Univ. of Bristol, Bristol, United Kingdom), and Bruce W. Drinkwater (Dept. of Mech. Eng., Univ. of Bristol, University Walk, University Walk, Bristol BS8 1TR, United Kingdom, b.drinkwater@bristol.ac.uk)

Acoustic waves with screw dislocations at their wavefronts, or acoustic vortices, are characterized by an azimuthal phase dependence. The time average of an acoustic vortex forms a circular region of high pressure which can be used to trap particles, and by manipulating the vortex axis, it can controllably translate them. In addition, acoustic vortices carry orbital angular momentum which can be transferred to absorbing particles causing them to rotate. This paper describes various new observations of these phenomena. In the first example, microparticles are manipulated in a host liquid at high ultrasonic frequencies (2 MHz), and a controlled translation and rotation is demonstrated. In this viscously dominated system, small particles rotate slowly with the liquid and larger particles are drawn into the centre of the vortex. The particle dynamics are explained as a fine balance between the transfer of angular momentum and the action of acoustic radiation forces. In the second example, much larger particles are levitated in air in the low ultrasonic range (40 kHz) and controlled translation and rotation is again observed. Here, viscosity is shown to play a lesser role in the particle dynamics, and conditions are observed under which the particles are rapidly ejected from the vortex core.

9:00

3aPA5. Induced radiation force and torque on a viscoelastic particle in an ideal fluid. José P. Leao and Glauber T. Silva (Phys., Federal Univ. of Alagoas, Av. Lourival Melo Mota, sn, Maceio, Alagoas 57035-557, Brazil, glauber@pq.cnpq.br)

The interaction of an acoustic wave with a suspended particle may produce radiation force and torque on the particle through linear and angular momentum transfer. Theoretical analysis of the radiation force exerted on a rigid, compressible fluid, elastic solid, and layered spherical particle is abundantly found in the literature. Nevertheless, less attention has been devoted to cases involving a homogeneous viscoelastic particle, even though viscoelastic materials are ubiquitous. Here, the radiation force and torque exerted on this kind of particle is studied in detail using the fractional Kelvin-Voigt viscoelastic model. Analytical expressions are obtained in the monopole-

3a WED. AM

dipole approximation to a small particle in the long-wave limit. Considering a traveling plane wave interacting with a high-density polyethylene (HDPE) particle, we found that the axial radiation force is negative, i.e., the force and the wave propagation are in opposite directions. For a first-order Bessel vortex beam, a full 3D tractor beam develops on the HDPE particle placed in the beam's axis. Furthermore, negative axial radiation torque also appears on the particle, i.e., the torque and the beam's angular momentum are contrariwise. In addition, possible implications of this method to the advancement of acoustophoretic techniques will be outlined.

9:15

3aPA6. A modal analysis view of minimum phase response of acoustic enclosures. Sahar Hashemgeloogerdi and Mark F. Bocko (Elec. and Comput. Eng., Univ. of Rochester, 405 Comput. Studies Bldg., Rochester, NY 14627, shashemg@UR.Rochester.edu)

When the response of an acoustic system is minimum phase, the existence of a stable and causal inverse is assured, which enables compensation for acoustic effects such as reverberation. In this paper, we explore the ranges of minimum phase behavior of one, two, and three-dimensional acoustic enclosures in light of their modal frequency response. We show that a one-dimensional acoustic enclosure, which by definition contains only axial modes, is always minimum phase. The responses of two and three-dimensional acoustic enclosures are characterized by a combination of normal modes in multiple dimensions. We show that two and three-dimensional acoustic enclosures demonstrate minimum phase behavior over limited frequency intervals. Minimum phase response tends to be confined to low frequencies, and a cutoff frequency below which the enclosure is minimum phase is identified.

9:30

3aPA7. A computationally efficient method for the frequency-domain analysis of visco-thermal acoustic propagation in arbitrary geometries. Gustavo Martins and Julio A. Cordioli (Mech. Eng. Dept., Federal Univ. of Santa Catarina, Campus Universitário, Trindade, Florianópolis, SC 88040-970, Brazil, julio.cordioli@ufsc.br)

The traditional approach used to represent the visco-thermal acoustic propagation within arbitrary geometries is the numerical solution of the full linearized Navier-Stokes (FLNS) equations. The FLNS model, usually solved by means of the Finite Element method (FEM), requires a large number of DOFs to be considered at each node. Therefore, the FLNS becomes much more computationally expensive than the standard acoustic model. An alternative is the sequential linear Navier-Stokes (SLNS) model, which can reduce the computational cost of the solution. However, the SLNS can be still computationally expensive for some types of analysis such as optimization's procedures. The SLNS model is obtained by means of some simplifications and decoupling of the FLNS equations and requires the solution of a viscous and a thermal dimensionless scalar field. The computational cost of the SLNS approach is manly due to the requirement of the solution of these scalar fields at each discrete frequency. In this paper, a semi-analytical solution of these parameters is proposed and tested in acoustic problems with arbitrary geometries. The new approach shows precision similar to the SLNS model, while the computational cost is only slightly higher than the standard acoustic model.

9:45

3aPA8. The scattering of sound by a buried obstacle in an extended reaction ground. Yiming Wang and Kai Ming Li (Mech. Eng., Purdue Univ., 177 South Russell St., West Lafayette, IN 47907-2099, wang1679@purdue.edu)

The boundary integral equation (BIE) method is widely used in outdoor sound scattering problems due to its computational efficiency when compared with the finite element method. The advantage of the BIE is apparent for the case when the scattering surface is much smaller than the computational domain of the problem. However, one of the major difficulties of the BIE formulation is the need to obtain the required Green functions for a sound source placed in the vicinity of the extended surface—both above and below the air/ground interface. There have been some recent developments where accurate and fast computations of these Green functions become available for use in the BIE formulation. This paper discusses some of these accurate asymptotic solutions and addresses the required steps to develop the BIE formulation. The computation of sound fields for the obstacle buried partially/totally in an extended reaction ground are presented.

10:00–10:15 Break

10:15

3aPA9. Turbulence effects on broadband pulses propagating near the ground. D. Keith Wilson, Vladimir E. Ostashev (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Sandra L. Collier, Jericho E. Cain (U.S. Army Res. Lab., Adelphi, MD), and Sylvain Cheinet (French-German Res. Inst. of Saint-Louis, Saint Louis, France)

Broadband sound pulses (such as from gunfire and explosions) distort randomly as they propagate through atmospheric turbulence. Recently, a general theory was formulated to describe the statistical moments, including the mean and space-time coherence, of such signals propagating along line-of-sight paths (Ostashev *et al*., J. Acoust. Soc. Am. **136**(5), 2414–2431 (2014)). This presentation focuses on some practical issues related to the physical interpretation of the theoretical predictions for the moments, such as what the theory predicts for pulse signals and turbulence regimes of typical interest, and how the behavior of individual pulse events is reflected in the averages that can be compared directly to the theory. In particular, we examine the conditions when the scattering is dominated by pulse *wander* (variations in arrival time) and when is it dominated by pulse *spread* (broadening of the energy in the impulse). Conversion of real-valued signals to analytic signals, and the averaging of these representations, is also discussed.

10:30

3aPA10. Wind turbine radiated acoustic signals—Propagation in a temporally and spatial variable marine meteorological boundary layer. Marshall H. Orr (College of Earth, Ocean and Environment, The Univ. of Delaware, PO Box 254, Bryantown, MD 20375, rubyspiral@gmail.com), Kenneth E. Gilbert, Xiao Di (National Ctr. for Physical Acoust., The Univ. of MS, University, MS), and Mohsen Badiey (College of Earth, Ocean and Environment, The Univ. of Delaware, Newark, DE)

The temporal and spatial properties of an acoustic field radiated from a wind turbine operating in a marine meteorological boundary layer and a spatial temporal variable impedance boundary will be quantified using paraxial equation based numerical simulations. A sound speed field will be constructed using wind field measurements made with a 915 MHZ radar wind speed sensor and local temperature and humidity measurements obtained from a meteorological tower. Temporally variable impedance boundary conditions for a tidal marine marsh will be included in the simulations.

10:45

3aPA11. Infrasonic wind noise reduction comparison for wind fences and porous domes. JohnPaul Abbott (NCPA and Dept. of Phys. and Astronomy, Univ. of MS, 122 PR 3049, Oxford, MS 38655, jrabbott@go.olemiss.edu), Richard Raspet (NCPA and Dept. of Phys. and Astronomy, Univ. of MS, University, MS), John Noble, W. C. K. Alberts, and Sandra Colliler (U.S. Army Res. Labs., Adelphi, MD)

This paper reports on an investigation directly comparing the measured wind noise and detected acoustic signals for two types of co-located infrasonic wind noise reduction barriers. The first type is a set of cylindrically shaped wind fence enclosures and the second type is a set of 2.0 m diameter semi-porous fabric domes. The fence configurations included variations to the height, width, and number of filtering layers. Forty and fifty-five percent porosities were used for each of these configurations. Wind Noise reductions for the domes and the 5 m diameter fence are comparable at low wavenumbers, while the reductions for the 5 m fence is better at high wavenumbers. The porous domes and the 6 m high wind fence achieve comparable maximum reduction levels; however, the 6 m high fence mitigates noise for a broader wavenumber range. The 10 m diameter fence achieves the best reduction levels overall, while the porous domes are better at mid-range wavenumbers. Additional filtering layers improve all reductions and show the greatest improvement for the fences and the higher porosity dome.

11:00

3aPA12. Influence of source motion, wind, and temperature profiles on the effective impedance of an absorptive surface. Kai Ming Li and Bao N. Tong (Mech. Eng., Purdue Univ., 140 South Russell St., West Lafayette, IN 47907-2031, mmkmli@purdue.edu)

This paper presents a theoretical study of the sound field due to a moving source placed above an absorptive surface in a stratified medium in the presence of wind and sound speed profiles. The current analysis starts with the general equations for an isentropic inviscid flow field. A monopole source is assumed to be traveling parallel to the absorptive surface at a constant speed. The standard Lorentz transform can be applied and the Fourier decomposition can be used to express the sound field in an integral form that is amenable to further mathematical treatments. Either the fast field computation or the method of steepest descent can be used to evaluate the integral. A two-dimensional formulation was considered initially, but it can subsequently be extended to three-dimensional flow fields. The effect of source motion, wind, and temperature profiles on the acoustic properties of the absorptive surfaces have been explored.

11:15

3aPA13. Statistical moments of impulse propagation through the atmosphere. Jericho E. Cain, Sandra L. Collier (US Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, jericho.cain@gmail.com), Sylvain Cheinet (French-German Res. Inst. of Saint-Louis, Saint-Louis, France), Vladimir Ostashev, and D. K. Wilson (US Army Engineer Res. and Development Ctr., Vicksburg, MS)

In order to localize and classify propagating impulsive signals, a theoretical model that incorporates effects due to atmospheric turbulence, ground impedance, wind speed, wind direction, and refraction is needed to guide the analysis of data measured with acoustic sensing systems. Recently, measurements with planar arrays were made that form a database of acoustic measurements of impulse sources propagated through various atmospheric conditions, ranges, and ground types. This paper presents analyses of the statistical moments of this data for the purposes of comparison with recent theoretical models.

11:30

3aPA14. Empty cavity in a cavitating liquid: Features of flow structure. Valeriy Kedrinskiy (Physical HydroDynam., Lavrentyev Inst. of HydroDynam., Russian Acad. of Sci., Lavrentyev prospect 15, Novosibirsk 630090, Russian Federation, kedr@hydro.nsc.ru) and Ekaterina S. Bolshakova (Phys., Novosibirsk State Univ., Novosibirsk, Russian Federation)

Problem of empty cavity dynamics in a two-phase medium is considered. The initial equilibrium state of "cavity-medium" system is disturbed when pressure inside cavity falls abruptly up to 0. Rarefaction wave arising on an interface initiates a cavitation development. Two-phase mathematical model is applied to investigate the medium state dynamics. The medium parameters correspond to a distilled water state: microbubbles, 1.5 μm, their density 10^6 cm^{-3}, and gas concentration, about 10^{-5}. The numerical analysis has shown that interface "cavity-medium" becomes a cavitating spherical layer. The concluding process of cavity collapse can be characterized by two stages. First, the interface as a spherical layer in a result of its cumulation is transformed into a spherical bubbly cluster with 1 mm radius. Cluster contains $2.5\cdot10^5$ см$^{-3}$ microbubbles with 40 μm radii. Gas concentration is distributed from 20%,cluster center, up to 1% on its surface. Second, the cumulation of flow on the spherical bubbly cluster will determine a level of internal energy of compressed bubbly cluster and its further dynamics. The similar phenomenon was found in the experiments on the development, structure, and collapse of a rupture forming in cavitating layer of distilled water at its shock-wave loading. The analysis of experimental data has shown that a rupture in the cavitating layer is the cavity with interface as thin layer of cavitating liquid and its collapse tends to the bubbly cluster formation. [Support RFBR, grant 15-05-03336.]

11:45

3aPA15. Acoustic measurements of the noise generated by the Silver Fox Unmanned Aerial System. Frank S. Mobley (Human Effectiveness Directorate, U.S. Air Force, 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, frank.mobley.1@us.af.mil)

Acoustic measurements of the noise generated by the Silver Fox Unmanned Aerial System (UAS) were accomplished on a test fixture at Owens-Corning. These measurements, made in one-percent throttle increments, revealed a region over which the noise power curve was linear. Source noise directivity patterns were constructed for each throttle increment using a spherical harmonic series expansion and compared to directivity patterns constructed using a proposed linear interpolation methodology. Moreover, these predictions from the two source construction methods were compared to validation measurements and demonstrate that the interpolation method is viable for spherical harmonic source representations.

Session 3aPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Potpourri (Poster Session)

Frederick J. Gallun, Chair

National Center for Rehabilitative Auditory Research, VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 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

3aPP1. Hearing protection device field attenuation estimation systems. JR W. Stefanson (Auditory Protection and Performance Div., US Army Aeromedical Res. Laboratory/ Hearing Ctr. of Excellence, 6901 Farrel Rd., Fort Rucker, AL 36362, earl.w.stefanson.ctr@mail.mil) and William A. Ahroon (Auditory Protection and Performance Div., U.S. Army Aeromedical Res. Lab., Fort Rucker, AL)

The United States Army Aeromedical Research Laboratory has conducted comparative, developmental, and evaluation studies for Hearing Protection Device Field Attenuation Estimation Systems (HPD FAES). The Army's interest in protecting Soldiers' hearing led to the development of the Communications Earplug (CEP), which couples communication systems with a foam earplug in an attempt to block hazardous noise while still passing to the user the clearest speech signal attainable. A recent developmental study addresses the need to fit-test the CEP when used with foam or custom earplugs in a quick and objective fashion using a field-microphone in real-ear (F-MIRE) procedure. Commercial systems have also been evaluated along with a clinical audiometer used by the Army Hearing Program for application as an HPD FAES. A select few FAESs were studied for their attenuation measurement accuracy as compared to the laboratory standard ANSI/ASA S12.6. Completed and ongoing studies will be presented on the FAESs' utility, cost, and effectiveness.

3aPP2. Characteristics of 40,000 calls to the National Hearing Test. Charles S. Watson (Res., Commun. Disord. Technol., Inc., CDT, Inc., 3100 John Hinkle Pl, Bloomington, IN 47408, watson@indiana.edu), Gary R. Kidd (Speech and Hearing, Indiana Univ., Bloomington, IN), Jill E. Preminger (Dept. of Otolaryngol. Head and Neck Surgery and Commun. Disord., Univ. of Louisville, Louisville, KY), James D. Miller, Daniel P. Maki, and Alex Crowley (Res., Commun. Disord. Technol., Inc., Bloomington, IN)

The National Hearing Test (NHT) is a telephone-administered screen for hearing loss (Watson *et al.*, J. Am. Acad. Audiol., 2012) that obtains thresholds for three-digit sequences in a noise background. The NHT has been validated by comparing threshold SNR values to mean pure-tone loss. During five weeks in 2014 the NHT was offered without charge to the general public. Over 40,000 calls were made after articles describing the test appeared in eight large-circulation and 19 smaller circulation newspapers, estimated to reach less than 10% of the US public. Among those who completed the test, 81% were estimated to have clinically significant loss in one or both ears. Call numbers suggested that 88% were made from landlines and the remainder from cell phones. Threshold SNRs for cell phones were about 1.0 dB higher than those for landline phones. Samples of the callers were later contacted through telephone and email surveys. Responses indicated a positive influence in terms of the likelihood of seeking further evaluation and obtaining hearing aids if advised to do so. [Research supported by the National Institute for Deafness and other Communication Disorders of the National Institutes of Health under award number 5R44DC009719.]

3aPP3. Objective evaluation of the acoustic properties of various types of chest pieces in the modern acoustic stethoscopes. Karolina M. Nowak (Dept. of Endocrinology, Ctr. of Postgraduate Medical Education, Ceglowska 80 St., Warsaw 01-809, Poland, karolina.nowak@ippt.pan.pl) and Lukasz J. Nowak (Inst. of Fundamental Technolog. Res., Polish Acad. of Sci., Warsaw, Poland)

The acoustic properties of a stethoscope are largely determined by the construction of its chest piece, which can either be open or closed with a diaphragm. Different solutions are offered on the market and advertised for their advantages in sound quality. However, no objective data, neither supporting nor disproving the benefits resulting from implementing the specific features, are available. The aim of the present study is to provide such data. A laboratory stand for measuring velocities of vibrations of different points of diaphragms or skin surface during the actual auscultation examination was developed and constructed. Those vibrations are the primary source of the acoustic signal in a stethoscope, and thus, the obtained results provide important conclusions regarding the reasonability of using various types of terminals in chest pieces. It is shown, that thin, stiff diaphragms mounted on susceptible suspension rings ensure significantly better acoustic performance than the other investigated solutions. The influence of the shape and construction of the chest piece on the acoustic properties were evaluated during independent, complementary research, conducted using precision microphone placed in the earpiece of a stethoscope. The obtained results deny the common opinions regarding advantages of the bell-type chest pieces in low-frequency acoustic band.

3aPP4. On material properties and damping models for the dynamic modeling of the human middle ear by means of the Finite Element Method. Felipe S. Pires, Diego C. Arellano (Mech. Eng. Dept., Federal Univ. of Santa Catarina, Campus Universitário, Trindade, Florianópolis, SC 88040-970, Brazil, felipesmp.emc@gmail.com), Stephan Paul (Mobility Eng. Dept., Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil), and Julio A. Cordioli (Mech. Eng. Dept., Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil)

An accurate dynamic model of the human middle ear is a valuable tool to better understand the mechanisms involved in the human hearing and some middle ear pathologies. It is also essential to the design and evaluation of implantable hearing devices, which may be connected to the middle ear structures. A considerable number of studies describe the development and validation of such models, and a review of these studies shows that there is a significant spread of material and dynamic properties used in the models. In this work, a detailed Finite Element model of the human middle ear, including tympanic membrane, complete ossicular chain, joints, and soft tissues (ligaments and tendons), is constructed and used to assess the influence of the input properties in the middle ear transfer functions. A frequency-dependent acoustic impedance at the oval window is used to represent the

cochlea. Ossicle motion predicted by the model is compared with experimental data from the literature, and the effects of different material properties and damping models are investigated. Although variations in material properties can have an impact in the middle ear transfer functions, different damping models shown to have a much larger effect, possibly leading to considerable errors in the numerical predictions.

3aPP5. Sound-induced flash illusion revisited. Aya C. Kito, Takafumi Furuyama, Kohta I. Kobayasi, Shizuko Hiryu, and Hiroshi Riquimaroux (Graduate school of life and medical Sci., Doshisha, 1-3, Tatara Miyakodani, Kyotanabe, Kyoto 6100394, Japan, dmp1011@mail4.doshisha.ac.jp)

Auditory-visual illusion occurs when visual information and auditory information integrates in the brain. This is conceived because auditory information has temporal superiority than visual information. Purpose of the study is to investigate mutuality influence of auditory information and visual information. Subjects answered the number of sounds and flashes when the auditory and visual stimuli are presented. We selected five auditory and visual stimuli (percentage answered as 2 times were 0%, 25%, 50%, 75%, and 100%) from the auditory only and visual only case and combined the stimuli. Subjects answered the number of sounds and flashes when the stimuli are presented simultaneously. When comparing the rate of unilateral and bilateral case, auditory only case and a combination of ambiguous auditory stimuli and definite visual stimuli, auditory information tends to shift toward visual information. This result suggests that in this case, visual information has temporal superiority so auditory information can perceive slight difference because it is influenced by definite visual information and flash-induced sound illusion occurs. From this result, we can suggest that temporal illusion occurs from temporal ambiguity of stimuli rather than temporal resolution of modality.

3aPP6. Effects of aging on auditory duration discrimination. Rachael Luckett and Edward L. Goshorn (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, Hattiesburg, MS 39406-0001, rachael.luckett@eagles.usm.edu)

The effects of aging on auditory processing are well documented. The involved anatomy includes the entire auditory pathway from peripheral to central mechanisms. Existing procedures to identify auditory processing disorders include a wide variety of signals with distortions or reduced redundancy in the temporal or frequency domains or the addition of noise. This project investigated the effects of aging on the ability to discriminate duration of a musical auditory signal. A musical signal was used to add temporally-variant acoustical information that would not be present in a pure tone. A digitized.wav piano note (f_o = 422 Hz) with gradually decaying amplitude was edited in Sound Forge to produce two signals equal in spectrum, RMS amplitude, and VU level but differing durations (1000 and 800 ms). One hundred pairs of these musical signals were arranged in random order with half equal and half unequal in duration. The subject's task was to designate same or different. Subjects ranged in age from 18 to 90 years. A regressions analysis revealed that age is a significant (p = .002) coefficient for predicting performance on a duration discrimination task. The findings suggest that duration discrimination tasks may contribute to identification of auditory processing disorders in adults.

3aPP7. Statistical modeling of expected rates of permanent hearing loss in newborn infants with data derived from the center for disease control. Edward L. Goshorn, Charles G. Marx, Kimberly Ward, and Marietta Paterson (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, PsychoAcoust. Res. Lab., Hattiesburg, MS 39401, edward.goshorn@usm.edu)

The Center for Disease Control (CDC) began gathering data on Universal Newborn Hearing Screening (UNHS) in 1999 and had data from all states and territories by 2005. The purpose of UNHS is to identify permanent hearing loss (PHL) in infants and make referrals for intervention. Health-care professionals may use CDC data to monitor the effectiveness of screening/diagnostic programs. Effective monitoring may designate whether or not an "expected" number of infants has been identified for a given birth rate. However, because the incidence of PHL in newborn infants is so low (nine-year CDC average across all states and territories equals 0.0013), a relatively large number of infants may be screened before the first occurrence of PHL. Therefore, there is legitimate concern that true positives will be missed due to a high rate of true negatives. Clinically useful models for comparing observed to expected data are needed. CDC data from 2005 to 2013 were used to set boundaries for binomial and negative-binomial distributions. These derived distributions were used to produce tables/graphs showing expected values and confidence intervals for an appropriate range of incidences and quantity of infants screened. The authors also offer actions to pursue if observed data vary significantly from models.

3aPP8. Phonetically balanced and psychometrically equivalent monosyllabic word lists for word recognition testing in Thai. Sajeerat Poonyaban, Pasinee Aungsakulchai, Charturong Tantibundhit (Dept. of Elec. and Comput. Eng., Faculty of Eng., Thammasat Univ., Khlong Luang, Pathumthani, Thailand), Chutamanee Onsuwan (Dept. of Linguist, Faculty of Liberal Arts, Thammasat Univ., Dept. of Linguist, Faculty of Liberal Arts, Thammasat University, Rangsit Campus, Khlong Luang 12120, Thailand, consuwan@hotmail.com), Rattinan Tiravanitchakul (Dept. of Commun. Sci. and Disord., Faculty of Medicine, Mahidol Univ., Ratchathewi, Bangkok, Thailand), Krit Kosawat (National Electronics and Comput. Technol. Ctr., Khlong Luang, Pathumthani, Thailand), and Adirek Munthuli (Dept. of Elec. and Comput. Eng., Faculty of Eng., Thammasat Univ., Khlong Luang, Pathumthani, Thailand)

In speech audiometry, a common and crucial method to obtain a supra-threshold (dB) at which words are repeated with maximum accuracy is referred to as word/speech recognition testing. For Thai, Thammasat University and Ramathibodi Hospital Phonetically Balanced Word Lists 2015 (TU-RAMA PB'15) were created with five lists, each with 25 monosyllabic words. Besides its phoneme distributions being based on large-scale Thai spoken corpora [1], TU-RAMA PB'15 is in line with TU PB'14 [2], [3] with emphasis on phonetic balance, symmetrical phoneme occurrence, and word familiarity. To evaluate its homogeneity in terms of decibel intelligibility, the lists were recorded and presented to 10 normal hearing participants, ranging from 0 to 50 dB HL in 2 dB increments (ascending order) until they repeated correct verbal responses. Using logistic regression, regression slopes and intercepts were calculated to estimate percentage of correct performance at any given intensity and to construct psychometric functions for every list. Derived psychometric function slopes ranged from 0.2015 to 0.2262 while intensities required for 50% intelligibility ranged from 17.0876 to 20.8856. Two-way Chi-Square analysis performed on both parameters indicated that there was no significant difference among the five lists. Further testing is needed to verify the use among the hearing-impaired individuals.

3aPP9. Prevalence of acoustic reflexes in the United States. Gregory A. Flamme, Kristy K. Deiters, Stephen M. Tasko (Speech Pathol. and Audiol., Western Michigan Univ., 1903 W. Michigan Ave., MS 5355, Kalamazoo, MI 49008, greg.flamme@wmich.edu), and William A. Ahroon (Auditory Protection and Performance Div., US Army Aeromedical Res. Lab., Fort Rucker, AL)

The acoustic reflex is a contraction of the middle ear muscles in response to high-level sounds. Acoustic reflexes are invoked as a protective mechanism in some damage-risk criteria (DRC). However, acoustic reflexes are not always observed among people without auditory dysfunction, and should not be included in DRC unless there is 95% certainty that 95% of the population have acoustic reflexes. In the current study, we present the prevalence of acoustic reflexes among people 12 years and older ($N > 11,400$), using data from the National Health and Nutrition Examination Survey (NHANES). The NHANES can be used to produce prevalence estimates generalizable to the non-institutionalized U.S. population. Ipsilateral reflexes were screened at two elicitor frequencies and detected using Frequentist methods and via Kalman filtering of the reflex trace. Reflexes are pervasive only among those with hearing thresholds better than 15 dB HL at all frequencies, and fall below the criterion certainty with poorer sensitivity even at lower frequencies. Age and tympanometric variables are also related to reflex detection. Reflex prevalence is generally high among young people with adequate hearing sensitivity for unrestricted military duty, but the prevalence is not uniform among audiometric configurations within this hearing profile.

3aPP10. Acoustic models of co-varying vocal roughness and breathiness. Mark D. Skowronski (Commun. Sci. and Disord., Univ. of South Florida, 4202 East Fowler Ave., PCD 3004, Tampa, FL 32620, skowronski@usf.edu), Lisa M. Kopf (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), Rahul Shrivastav (Office of the Vice President for Instruction, Univ. of Georgia, Athens, GA), and David A. Eddins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Dysphonia is characterized by several vocal qualities including breathiness and roughness. These qualities can coexist within a single voice. To examine the potential interaction between judgments of one voice quality in the presences of another, a large set of synthetic voices that co-varied along the breathiness and roughness continua were created and evaluated by 10 listeners. The perception of roughness (synthesized via amplitude modulation) was unaffected by degree of breathiness. The perception of breathiness, however, was affected by the degree of roughness: for low-breathy voices breathiness increased by about 2 dB as roughness increased from no roughness to maximum roughness; yet for high-breathy voices, breathiness decreased by about 2 dB as roughness increased. Current acoustic models of voice quality (pitch strength, auto-correlation peak, cepstral peak, partial loudness ratio, and glottal-noise-excitation ratio) do not explain the observed interaction of roughness and breathiness. To explain the observed interaction, we evaluate the possibility that roughness perception is dominated by temporal cues (which are unaffected by the presence of breathiness) and that breathiness perception is driven by spectro-temporal cues which are affected by the sub-harmonics generated by amplitude modulation. The sub-harmonics partially contribute to non-harmonic energy (increasing breathiness) and partially mask aperiodic energy (decreasing breathiness).

3aPP11. The effects of acoustic variability on absolute pitch categorization: Evidence of contextual tuning. Stephen C. Van Hedger, Shannon L. Heald, and Howard C. Nusbaum (Psych., The Univ. of Chicago, 5848 S. University Ave., Beecher 406, Chicago, IL 60637, stephen.c.hedger@gmail.com)

Absolute pitch (AP) is defined as the ability to label or produce a musical note without the aid of a reference note. Despite the large amounts of acoustic variability encountered in music, AP listeners generally experience perceptual constancy for different exemplars within note categories (e.g., recognizing that a C played on a tuba belongs to the same category as a C played on a piccolo). The present studies investigate whether AP possessors are sensitive to context variability along acoustic dimensions that are not inherently linked to the typical definition of a note category. In a speeded target recognition task, AP participants heard a sequence of notes and pressed a button whenever they heard a designated target note. Within a trial the sequence of notes was either blocked according to note-irrelevant variation or contained a mix of different instruments (Experiment 1), amplitude levels (Experiment 2), or octaves (Experiment 3). Compared to the blocked trials, participants were significantly slower to respond in the mixed-instrument and mixed-octave trials, but not the mixed-amplitude trials. Importantly, this performance difference could not be solely attributed to initial performance differences between instruments, amplitudes, or octaves. These results suggest that AP note identification is contextually sensitive.

3aPP12. Motor suppression of the auditory system extends to the brainstem frequency following response and is mediated by attentional demands. Serena Klos and Howard C. Nusbaum (Psych., The Univ. of Chicago, 5848 S University Ave., Chicago, IL 60637, sklos@uchicago.edu)

Neural theories of auditory perception often characterize subcortical structures as relay stations by which acoustic input is passively encoded into a representation that can be recognized by cortical networks. However, efferent projections throughout the peripheral auditory pathway (Huffman & Henson, 1990) suggest a corticofugal network consisting of ascending and descending pathways may play an important role in the perception of acoustic signals. Based on cortical suppression evidence in the primary auditory cortex during movement (Schneider *et al*., 2014), we investigated whether similar suppressive effects can be seen at the level of the auditory brainstem and whether there are interactions with attention. The Frequency Following Response (FFR) to a 440Hz sine tone was measured while participants engaged in two finger tapping tasks equated in motor behavior but varying in attentional demand. Spectral peak analysis of the FFRs revealed decreased amplitude at 440 Hz for the tapping task that required more attention, suggesting that rather than a general suppression of the peripheral auditory pathway during motor behavior, the interaction between the motor system and the auditory brainstem is mediated by attentional networks that are involved in allocating resources to various sensory networks. [This work was supported in part by ONR grant DoD/ONR N00014-12-1-0850.]

3aPP13. Cortical dynamics of spatial and non-spatial auditory selective attention. Yuqi Deng, Hannah Goldberg, Barbara Shinn-Cunningham, and Inyong Choi (Boston Univ., 677 Beacon St., CompNet, Boston, MA 02215, vydeng@bu.edu)

Auditory selective attention suppresses processing of task-irrelevant stimuli, and it is crucial for effective communication in social settings. Previous studies showed that space- and pitch-based auditory attention engages different neural networks. However, the cortical dynamics underlying spatial and non-spatial auditory attention are unclear. Since accumulating evidence suggests that selective suppression is related to alpha band oscillation (8–14 Hz), we examine the spatial and non-spatial attentional modulation of alpha oscillation power as well as event-related potential (ERP) and behavioral performance. Using Electroencephalography (EEG) in humans, we compare behavior and physiological measures during *focused* attention (where listeners maintain focus on one "target" stream) and *broad* attention (where listeners are prepared to switch attention to a "super-target" stream which may or may not appear after the "target" stream) in spatial and non-spatial settings. We find that spatial attention shows overall stronger alpha power and different distribution in the central-parietal cortex. We also find that the monitoring cost of *broad* attention is higher in spatial attention, which is supported by a stronger ERP modulation. Our findings help elucidate the cortical dynamics involved in spatial and non-spatial auditory attention.

3aPP14. Age-related differences in auditory cortical representations of spatial cues. Erol J. Ozmeral, David A. Eddins, and Ann C. Eddins (Commun. Sci. and Disord., Univ. of South Florida, 3802 Spectrum Blvd., Ste. 210, Tampa, FL 33612, eozmeral@usf.edu)

Converging evidence indicates that binaural processing declines with age and is impaired further by age-related hearing loss. Research in young, normal-hearing adults indicates that spatial coding is governed by an opponent-channel (OC) mechanism. Under the OC model, the sensitivity to changes or shifts in perceived lateralization can be predicted based on the direction and magnitude of the shift, where large, outward shifts are predicted to cause a greater neural response than small and/or inward shifts. It is possible that age-related changes in spatial coding reflect changes in the OC mechanism, perhaps due to reduced neural inhibition or a more general reduction in temporal precision. Cortical event-related responses elicited by both ITD and ILD changes were measured using a 64-channel montage during a passive, continuous change-detection paradigm. Source localized cortical activation patterns were analyzed with regard to degree of lateralized shift, direction of shift (toward or away from perceived midline), and listener group. Results are consistent with an OC mechanism of spatial coding for both younger and older adults. Behavioral data from the same subjects are reported for correlation analyses with neural data. The present study provides a foundation for understanding the cortical dynamics of age-related changes in spatial tuning.

3aPP15. Cortical dynamics during spectrotemporal processing as indexed by directed functional connectivity. Ann C. Eddins and David A. Eddins (Commun. Disord. & Sci., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, aeddins@usf.edu)

Neural processing in the cortex plays a key role in our ability to discriminate changes in acoustic features important for understanding speech. Converging evidence demonstrates functional asymmetries between hemispheres such that the left hemisphere (LH) typically shows greater sensitivity to temporal changes while the right hemisphere (RH) shows greater sensitivity to spectral change. Most human studies to date have evaluated static differences in hemispheric activation with little attention given to potential dynamic changes within and across hemispheres during stimulus processing. The present study quantifies changes in the cortical representation of spectral (0.5 and 2.0 cycles/octave), temporal (2, 8, 32, and 64 Hz), and spectro-temporal (combination of both) modulation frequency. Cortical event-related responses were measured using a 64-channel montage during a passive, continuous change-detection paradigm in young, normal-hearing listeners. Individual and group data were preprocessed and source-localized for all conditions, with regions of interest (ROI) defined on group data. To determine whether activity in one ROI dynamically influences another during stimulus processing, Granger causality modeling was used to estimate directed functional connectivity. Preliminary results show robust directional processing that may originate in one hemisphere (i.e., spectral processing in RH or temporal processing in LH) but dynamically engages cortical regions within and across hemispheres.

3aPP16. The initial head gesture and coordinate system in anthropometric parameters measurement for head-related transfer functions customization. Guangzheng Yu, Yingyang He, and Bosun Xie (South China Univ. of Technol., Wushan Rd. 381#, Tianhe District, Guangzhou, Guangdong 510640, China, scgzyu@scut.edu.cn)

Head related transfer functions (HRTFs) depend on the sound source position and individualized anthropometry of subject. Accordingly, HRTF measurement usually accompanies with head and pinna-related anthropometric measurement on corresponding subjects. Based on a baseline database of HRTFs and anthropometric parameters, individualized HRTFs of new subject can be customized from anthropometric measurement (Zotkin, *et al*., 2003). Obviously, the accuracy of anthropometric measurement influences the performance of customized HRTFs. However, the measurement of some pinna-related anthropometric parameters, such as the pinna rotation angel and pinna flare angle, is sensitive to the initial head gesture and coordinate system. In present work, the Frankfurt plane was adopted to initialize the head gesture under the same coordinate system so as to ensure the consistency of anthropometric and HRTF measurement. The suggested method is adopted to measure the head and pinna-related anthropometric parameters of 56 subjects, yielding reasonable and consistent results. [This work was supported by the National Natural Science Foundation of China, Grant No. 11104082.]

3aPP17. Front-and-Overhead Energy Ratio and immersive sound field rendering with height channels. Sungyoung Kim and Mark J. Indelicato (ECT Eng. Technol., Rochester Inst. of Technol., ENT-2151, 78 Lomb Memorial Dr., Rochester, NY 14623, sungyoungk@gmail.com)

In a discrete channel-based music reproduction, height channels are required to completely manipulate increased immersiveness and enhanced appropriateness for a realistic three-dimensional (3D) sound field. Previous subjective evaluation results showed that the configuration of four height loudspeakers significantly changed perceived immersiveness and appropriateness, and the listeners preferred height-loudspeaker configurations with being “frontal” and “full.” Subsequent analysis failed to determine a prediction model that could account for the variation of the listeners’ perceptual responses using conventional physical parameters including inter-aural difference and coherence. In this paper, the authors proposed a new metric-the ratio between sound energy from front and overhead directions-as a physical parameter for the prediction model. A coincident pair of bidirectional microphones (one facing front and the other facing overhead) measured acoustical impulse responses (IRs) at the listening position for eight configurations of four height channels. Subsequent analysis measured the Front-and-Overhead Energy Ratio (FOER) between two IRs from the eight configurations and found that the ratios were highly correlated with the listeners’ subjective rank data on perceived immersiveness and appropriateness (r = -0.8037). The result implicated that the appropriated configuration of height channels is a critical factor to render a convincing and immersive 3D sound field due to its direct influence on the FOER.

3aPP18. Efficient algorithm and localization experiment on spherical microphone array recording and binaural rendering. Yu Liu, Bosun Xie, Haiming Mai, and Jiayan Chen (South China Univ. of Technol., Wushan Rd. 381., Tianhe District, Guangzhou 510640, China, phbsxie@scut.edu.cn)

Spherical microphone array recording and binaural rendering (SMABR) is a novel spatial sound technique, which records spatial information of sound field, transfers to dynamic binaural signals, and then renders via headphone. To improve the computational efficiency of SMABR, the present work proposes a PCA-based (principal component analysis) algorithm for dynamic binaural synthesis from the beamforming outputs of spherical microphone array, in which only 33 pairs of sharable filters are required. Incorporated the proposed algorithm, a PC-based SMABR system consisting of a 64-channel spherical microphone array, an electromagnetic head-tracker, and a headphone is established. A virtual source localization experiment is carried out to evaluate the system and algorithm. Results indicate that the system yields reasonable localization performance within a target virtual source region near the horizontal plane. However, larger localization error is also observed for high elevation above 30 degrees or low elevation below -30 degrees. Localization error at high or low elevation is caused by the spatial aliasing error of spherical microphone array at high frequency, which spoils the spectral cue for elevation localization. Therefore, further improvement on spherical microphone array recording is needed. [Supported by the National Natural Science Foundation of China, Grant No.11174087.]

3aPP19. Neural network based speech enhancement applied to cochlear implant coding strategies. Tobias Goehring (ISVR, Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, T.Goehring@soton.ac.uk), Federico Bolner (Cochlear Technol. Ctr. Belgium, Mechelen, Belgium), Jessica J. M. Monaghan (ISVR, Univ. of Southampton, Southampton, United Kingdom), Bas van Dijk (Cochlear Technol. Ctr. Belgium, Mechelen, Belgium), Jan Wouters (ExpORL, KU Leuven, Leuven, Belgium), Marc Moonen (ESAT, KU Leuven, Leuven, Belgium), and Stefan Bleeck (ISVR, Univ. of Southampton, Southampton, United Kingdom)

Traditionally, algorithms that attempt to significantly improve speech intelligibility in noise for cochlear implant (CI) users have met with limited success, especially in the presence of a fluctuating masker. Motivated by previous intelligibility studies of speech synthesized using the ideal binary mask, we propose a framework that integrates a multi-layer feed-forward artificial neural network (ANN) into CI coding strategies. The algorithm decomposes the noisy input signal into time-frequency units, extracts a set of auditory-inspired features and feeds them to the ANN to produce an estimation of which frequency channels contain more perceptually important information (higher signal-to-noise ratio, (SNR)). This estimate is then used accordingly to suppress the noise and retain the appropriate subset of channels for electrical stimulation, as in traditional N-of-M coding strategies. Speech corrupted by various noise types at different SNRs is processed by the algorithm and re-synthesized with a vocoder. Evaluation has been performed in comparison with the Advanced Combination Encoder (ACE) in terms of classification performance and objective intelligibility measures. Results indicated significant improvement in Hit—False Alarm rates and intelligibility prediction scores, especially in negative SNR conditions. Findings suggested that the use of ANNs could potentially improve speech intelligibility in noise for CI users and motivated subjective listening experiments that will be presented together with the objective results.

3aPP20. Effects of phase difference on the binaural perceiving intensity. Sanmun Kim and Young H. Kim (Appl. Acoust. Lab, Korea Sci. Acad. of KAIST, 105-47, Baegyanggwanmun-ro, Busanjin-gu, Busan 614-100, South Korea, physicool13@gmail.com)

It is well known fact that human localize the direction of sound source in transverse plane according to interaural time difference in low frequency and interaural level difference on high frequency. In this research, we tested possibility for interaural time difference to be interpreted as interaural level difference when it is perceived by brain. To test the hypothesis, we used sinusoidal wave and commercial music to study the effect of interaural time difference on the perception of loudness. Each sound sample was regulated to have interaural time difference and given to 9 subjects. Subjects were asked to answer which side was louder. For sinusoidal waves, more than 60% of subjects answered side with earlier sound arrival was louder. The effect of interaural time difference on interaural level difference was larger when it interaural time difference was given to lower frequencies. Music samples were regulated with 100 and 500 interaural time difference and 100% of subjects answered side with earlier sound arrival was louder. Results from the experiments imply the possibility for interaural time difference interpretation by interaural level difference on localization of sound sources.

3aPP21. Our musical brain: Uncovering the neurophysical activation mechanism behind human perception of acoustics. Garrett W. Arosemena Ott, Tyler Blazey, Anish Mitra, Abraham Z. Snyder, and Marcus E. Raichle (Washington Univ., 6515 Wydown Boulevard, St. Louis, MO 63105, g.ott@wustl.edu)

If, in fact, neuronal activity patterns in the brain are dominantly driven by external forces and environmental stimuli (Hasson *et al.* 2004), rather than individual variation, music, and certain musical features, should elicit directly correlated neuronal responses across specific structures of the brain—effects as strong as that examination and analysis of brain activity in one subject could accurately predict activations in another (brain). Importantly, locating areal responses and activity patterns to musical stimulation could uncover novel information as to brain network functions, as well as yield insight into the neural foundations of the creative mind. Subjects, comprised of highly trained (HT) classical musicians (St. Louis Symphony) undergo functional Magnetic Resonance Imaging (fMRI) blood-oxygenation-level-dependent (BOLD) scanning to map neuronal activity triggered by intervals of music (Haydn Symphony No. 3), silence, and noise. Observed activation included the auditory, posterior cingulate, and visual cortices. The posterior cingulate cortex, a central node in the brain's Default Mode Network, has been strongly implicated in associations with episodic memory retrieval, working memory performance, human awareness, and several intrinsic control networks. The visual cortex is responsible for processing visual information; such activation may affirm the neuro-physical manifestation of HT performers' staple sensualization of music—music experienced as color, flavor, feeling, mood, etc.

3aPP22. Robust analysis of sound field reproduction by ambisonics based on singular-value decomposition. Dan Rao and Bosun Xie (Acoust. Lab.,School of Phys.,South China Univ. of Technol.,Tianhe district, Guangzhou, Guangdong 510641, China, phdrao@scut.edu.cn)

Ambisonics is a series of spatial sound systems with flexible loudspeaker configuration, which aims at reconstructing physical sound field in local region. Robustness of Ambisonics reproduction, which reflects the sensitivity of reproduced sound field to small errors, such as loudspeaker gain mismatch, is an important performance of sound reproduction. Condition number is generally used as an index to evaluate the robustness on whole, but this is insufficient because the robustness of reproduced sound field may be related to the target reproduction direction. To gain insights into the detail feature of robustness in Ambisonics reproduction, a method based on singular-value decomposition (SVD) is proposed to analyze reproduced sound field. The results indicate that the small singular values of transfer matrix and the proportion of Ambisonics coded vector projecting to the corresponding singular vectors dominate the robustness of Ambisonics reproduction. The robustness becomes worse only when such proportion of Ambisonics coded vector projection have relatively large energy. The proposed method is validated by simulating the sound pressure errors due to random mismatch of loudspeaker signals gain in some horizontal and spatial loudspeaker configurations. [Work supported by the National Natural Science Foundation of China, Grant No.11174087, and the State Key Lab of Subtropical Building Science, South China University of Technology, Grant No. 2014KB23.]

3aPP23. Interaural time and level interaction under free-field conditions. Brad Rakerd, Eric J. Macaulay, and William M. Hartmann (Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, rakerd@msu.edu)

Listeners judged the azimuthal location of sine tones presented in free field from an array of 25 small loudspeakers uniformly spaced over 180 degrees in the forward half of the horizontal plane. Probe microphones in the listener's ear canals recorded the tones, and transaural synthesis was used to reverse the interaural time difference (ITD) or the interaural level difference (ILD) across the mid-sagittal plane. Alternatively, the synthesis maintained the natural, consistent ITD and ILD. In the frequency range of 1000 Hz and below, localization judgments reflected the ITD, even as the interaural phase difference (IPD) surpassed 180 degrees with increasing azimuth or frequency, as long as the ILD and ITD had the same sign. In the reversed condition, and with small IPD, judgments reflected a compromise between ITD and ILD, with the ILD weight increasing with increasing frequency. But when the IPD exceeded a critical angle (about 120 degrees of phase) the influence of the ITD dramatically changed: Above 500 Hz, the localization judgments were consistent with ITD from a slipped cycle. At 500 Hz, judgments became chaotic. No critical angle occurred at 250 Hz. These conclusions are relevant in assessing the roles of interaural differences in sound localization. [Work supported by the AFOSR.]

3aPP24. The test-retest reliability of measurement procedures on most comfortable loudness levels for pure-tones. Cheng-Yu Ho (Holistic Education Ctr., Mackay Medical College, No. 46, Sec. 3, Zhongzheng Rd., Sanzhi Dist., New Taipei City 25245, Taiwan, swellfishyu@gmail.com), Pei-Chun Li (Dept. of Audiol. and Speech-Lang. Pathol., Mackay Medical College, New Taipei City, Taiwan), and Shuenn-Tsong Young (Holistic Education Ctr., Mackay Medical College, New Taipei City, Taiwan)

This study aimed to evaluate test-retest reliability of the forced-choice paired-comparison measurement procedures on most comfortable loudness levels (MCL) for pure-tones. Previous studies indicated that the MCL for pure-tones may be a range of level instead a fixed level, since the fixed-level MCL tends to be more inconsistent than a range of MCL. The ascending and descending measurement procedures were mostly used in the MCL for pure-tones, however the lower MCL obtained from ascending procedure, and higher MCL got from descending procedure, and since the low test-retest reliability of ascending and descending procedures, there are no well-established measurement procedures of MCL for pure-tones. The ascending and descending forced-choice procedure of MCL for speech might provide respectively 83% and 84% test-retest reliability, but the test-retest reliability measurement procedures of MCL for pure-tones have not been known. Therefore, this study proposed a new forced-choice paired-comparison measurement procedure on MCL for pure-tones, and evaluated the test-retest reliability of this proposed procedure. Normal hearing subjects are recruited to conduct this experiment, and evaluated by Pearson correlation. This experiment is ongoing, and the data collection would be finished before the presentation on the 170$^{\text{th}}$ Meeting of the Acoustical Society of America.

Session 3aSA

Structural Acoustics and Vibration, Signal Processing in Acoustics, and Physical Acoustics: Nonlinear Techniques for Nondestructive Evaluation

Brian E. Anderson, Chair
Dept. of Physics and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602

Invited Papers

8:00

3aSA1. Stress and strain monitoring in metals and alloys by guided acoustic waves with the aid of anharmonic effects and stress stiffening. Julian Grill and Wolfgang Grill (ASI Analog Speed Instruments GmbH, Burgweg 8, Koenigstein im Taunus, Hessen 61462, Germany, jg@analogspeed.de)

The basic theoretical features including anharmonic effects allowing structural health monitoring and the detection of stress and strain by guided acoustic waves are exemplified. Experimental results for stress and strain monitoring covering the linear elastic regime and the regime of plastic deformation are demonstrated for brass, copper and a standard aluminum alloy. Practically oriented applications, including also the monitoring of structural defects, load, weakening of structural components by overload and vibrations, and the detection of temperature to compensate for temperature dependent effects influencing structural health monitoring by acoustic waves, are presented for applications involving sections and components of aircrafts.

8:20

3aSA2. Nonlinear ultrasonic waves for structural monitoring: Thermal stress measurement and guided wave management. Francesco Lanza di Scalea (Structural Eng., Univ. of California San Diego, 9500 Gilman Dr., MC 0085, La Jolla, CA 92093, flanzadi@ucsd.edu), Claudio Nucera (Deutsche Bank, La Jolla, CA), and Simone Sternini (Structural Eng., Univ. of California San Diego, La Jolla, CA)

This presentation will cover two aspects of ultrasonic nonlinear wave propagation in solids. First, a new model is proposed to justify the existence of wave nonlinearities in constrained solids subjected to thermal excursions. This problem is solved on the basis of the interatomic potential of the solid that indicates a “residual” strain energy, due to the prevented thermal expansion, that is at least cubic as a function of strain. This study finds applications in the monitoring of thermal stresses in buckling-prone structures, such as continuously welded railroad tracks and pipelines. Experimental tests conducted on railroad tracks with realistic support will be also presented. Second, in the case of waveguides the efficiency of nonlinear ultrasonic testing based on higher-harmonic generation strongly relies on the correct identification of favorable combinations of primary and resonant double-harmonic nonlinear wave modes. This presentation will identify these combinations of wave modes in complex waveguides by extending the classical Semi-Analytical Finite Element formulation to the nonlinear regime, and implementing it into a highly flexible commercial Finite Element code. The proposed algorithm is benchmarked for four case-studies, including a railroad track, a viscoelastic plate, a composite quasi-isotropic laminate, and a reinforced concrete slab.

8:40

3aSA3. Nonlinear time reversal signal processing techniques applied to acousto-mechanical imaging of complex materials. Serge Dos Santos (Inserm U930 “Imaging and Brain, INSA Ctr. Val de Loire, 3, Rue de la Chocolaterie, Blois, Centre-Val de Loire F-41034, France, serge.dossantos@insa-cvl.fr), Zuzana Dvorakova (Inst. of ThermoMech. AS CR, Prague, Czech Republic), Michael Caliez (LMR, INSA Ctr. Val de Loire, Blois, France), and Zdenek Prevorovsky (Inst. of ThermoMech. AS CR, Prague, Czech Republic)

Recent ten years have seen considerable development of experimental techniques for improving nonlinear NDT methods and harmonic imaging derived from Nonlinear Elastic Wave Spectroscopy (NEWS). Diagnostic ultrasonic imaging based on higher harmonics yields, among others, a better resolution in view of the decreased wavelength in comparison with the fundamental. Furthermore, using symmetry invariance, nonlinear Time Reversal (TR) and reciprocity properties, the classical NEWS methods are supplemented and improved by new excitations having the intrinsic property of enlarging frequency analysis bandwidth and time domain scales. The purpose of this paper is to present the extension of TR-NEWS for skin aging characterization using the Nonlinear Time Reversal signal processing tool known to localize, in a complex medium, sources of nonlinearity potentially responsible of complex material aging. Linear and nonlinear behavior of skin elasticity is measured locally thanks to an acousto-mechanical loading of the skin conducted with INSTRON loading machines specifically optimized for biomaterials. Hysteresis behavior coming from the complex loading of the skin has been identified with PM-space statistical approach, usually associated to aging process in NDT. Phenomenological hysteretic parameters extracted will be presented and associated to standard parameters used for skin characterization.

9:00

3aSA4. Detection and localization of microcracking in polymer concrete using coda wave interferometry (CWI) at resonance. Charfeddine Mechri, Mourad Bentahar (Acoustics, LAUM, Laboratoire d'acoustique de l'université du Maine, Ave. Olivier Messiaen, Le Mans 72000, France, charfeddine.mechri@univ-lemans.fr), Souad Toumi (Wave & Acoust., Haouari Boumdiene Univ., Laboratoire de Physique des Matériaux, Algers, France), Fouad Boubneider (Wave & Acoust., Haouari Boumdiene Univ., Laboratoire de Physique des Matériaux, Algers, Algeria), and Rachid El Guerjouma (Acoutics, LAUM, Laboratoire d'acoustique de l'université du Maine, Le Mans, France)

Nonlinear resonance and Coda Wave Interferometry (CWI) proved their potential to detect the evolution of a structure, namely, in the case of damage. Nevertheless, these techniques do not allow localizing defects in a structure. CWI may allow this localization through heavy statistical algorithms. In this case, the main problem remains the real time monitoring on one hand. On the other hand, in the case of highly scattering media, the amplitude of the acoustic pulse may cause the Coda signal to be affected by conditioning/relaxation effects. Moreover, Coda signal is very sensitive to experimental conditions, namely, to temperature. In order to overcome these inconveniences, we propose a method based on a comparative study of Coda signal contents in rest state and under a weak (linear) vibration. In this communication, we report the results obtained on a polymer concrete specimen excited under different resonance modes and for different dispositions of the sample which defect is not isotropic on which we apply simultaneously through transmission experiments in order to monitor changes in the coda signal.

9:20

3aSA5. Nonlinear ultrasonic technique for closed crack detection. Kyung-Young Jhang (School of Mech. Eng., Hanyang Univ., 204 Eng. Ctr. Annex, 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, kyjhang@hanyang.ac.kr) and Hogeon Seo (Dept. of Mech. Convergence Eng., Hanyang Univ., Seoul, South Korea)

The detection of cracks at the early stage of fracture is important in industrial structures in order to guarantee their structural safety. Ultrasound has been widely utilized in the field of nondestructive testing of materials. However, most of these conventional methods using ultrasonic characteristics in the linear elastic region are mostly sensitive to opened cracks but much less sensitive to such closed cracks. The nonlinear ultrasonic technique (NUT) based on the contact acoustic nonlinearity (CAN) has been considered as a promising method for the closed crack detection. However, most of the previous studies were limited to the modeling of the second-order harmonic wave generation at contacted interfaces and its verification by testing artificially contacted interfaces in the through-transmission method. In this study, we investigated experimentally the contact acoustic nonlinearity at a real crack by using the measurement system constructed in the reflection mode that permits the transducers to access the only single side of a test structure. Results showed that the magnitude of the second-order harmonic wave represented the existence of the closed area clearly and that the crack sizing performance was greatly improved by the combination of the linear and nonlinear ultrasonic techniques.

9:40–10:00 Break

10:00

3aSA6. Pulse inversion and scaling subtraction signal processing for nonlinearity based defect detection. Koen Van Den Abeele, Jan Hettler, Morteza Tabatabaeipour, and Steven Delrue (Dept. of Phys., KU Leuven - Kulak, E. Sabbelaan 53, Kortrijk 8500, Belgium, koen.vandenabeele@kulak.be)

When seeking out evidence for nonlinear behavior, various signal processing techniques can be applied for the comparison of two signals, one being a slight distortion of the other. For instance, the pulse inversion technique compares the responses to two out-of-phase excitation signals. Alternatively, one can compare the response at a finite (nonlinear) excitation amplitude to a scaled response at a very low (linear) excitation, as performed in the scaling subtraction technique. In this report, several examples are given in which these nonlinearity based signal processing techniques are used in practice to visualize damage features in solids. In view of kissing bond defect detection in friction stir welds, the pulse-inversion method was employed in a contact pitch-catch mode using a chirp signal. B-scan spectral heat maps obtained after pulse inversion allow to easily identify and size damage zones along the weld path. Second, the scale subtraction technique will be illustrated in combination with an ultrasonic sparse array SHM system to detect damage locations (impacts and delaminations in CFRP plates) without the knowledge of baseline signals taken on an intact specimen. Finally, we show that the phenomenon of Local Defect Resonance (LDR) can be facilitated and validated using the scaling subtraction technique.

10:20

3aSA7. Elasticity Nonlinear Diagnostic method for crack detection and depth estimation. Pierre-yves Le Bas, Brian E. Anderson, Marcel Remillieux (Geophys. group, EES-17, Los Alamos National Lab., MS D446, Los Alamos, NM 87545, pylb@lanl.gov), Lukasz Pieczonka (2. AGH Univ. of Sci. and Technol., Krakow, Poland), and Timothy J. Ulrich (Geophys. group, EES-17, Los Alamos National Lab., Los Alamos, NM)

In some Non-Destructive Evaluation techniques, detecting a crack is only the first step. Some applications like monitoring of used nuclear fuel canister also requires the characterization of the orientation of the cracks if any exist. Here, we show experimental results that expand the Time Reversal Elasticity Nonlinear Diagnostic method (TREND) to obtain the crack depth and orientation information. TREND is based on using time reversal to focus energy at a point of interest and then quantifying the nonlinearity at this point. By varying the frequency of the focal signal, one can interrogate different depth of the material, therefore getting access to information about the penetration depth of a crack. We will also show how Time Reversal can be used to select the orientation of the focus wave to any desired direction and how this, in turn, allows measuring different nonlinear responses according to this orientation and how this could be used to determine the orientation of cracks. [This work was supported by the US Department of Energy via the used fuel disposition campaign of the nuclear energy program.]

10:40

3aSA8. Model-based nonlinear guided wave approach for nondestructive evaluation in solid media. Younho Cho (School of Mech. Eng., Pusan National Univ., 10511, San 30, Jangjeon-dong, Geumjeong-gu, Busan 609-735, South Korea, mechcyh@pusan.ac.kr) and Weibin Li (Dept. of Aeronautics, Xiamen Univ., Xiamen, China)

The measurement of acoustic nonlinear response of guided wave propagation has been explored as a promising tool for early detection of micro-damages. Considering the high sensitivity of the nonlinear ultrasonic approach and the great advantages of the guided wave techniques, the nonlinear guided wave techniques have drawn significant attention. Frequency tuning and mode selection play a critical role for practical nonlinear guided wave testing. Among all the guided wave modes, it is of particular interest to find the more suitable modes for the improvement of efficiency. Analytical expressions and experimental observations of second harmonic generation of guided waves in an isotropic plate, are presented in this talk. Nonlinear parameters of guided waves are newly derived as a function of wave mode, geometric information of waveguides and frequency. Nonlinear features of various phase matched modes are discussed for the comparison of efficiency of second harmonic generation. The experimental results are consistent with the theoretical predictions. This study shows that physically based feature selection is essential for efficient generation of second harmonic guided waves. The feasibility of Lamb waves mixing for nonlinear ultrasonic test is also discussed in this work.

11:00

3aSA9. Soil-plate oscillator: A model for nonlinear acoustic landmine detection. Murray S. Korman (Dept. of Phys., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu) and James M. Sabatier (National Ctr. for Physical Acoust., Oxford, MS)

A soil-plate oscillator (SPO) apparatus involves a cylindrical column of granular medium (sand, soil, and pebbles) supported by a thin circular elastic plate (acrylic) that is rigidly clamped to the bottom of a thick walled aluminum tube. The plate is air-backed. The soil column is driven from above by two subwoofers electrically connected to an amplified swept sinusoidal slowly varying chirp. In nonlinear tuning curve experiments, the resonant frequency decreases with increased amplitude—representing a softening in the nonlinear system. Hysteresis effects can be observed along with slow dynamic behavior (associated with mesoscopic nano-scale nonlinear elasticity). In two tone tests, numerous combination frequencies are observed in the surface vibration of the soil along with the sum frequency (typically 200 Hz) and second and higher harmonics. Soil surface vibrations and plate vibrations show similar effects when the soil layer is even 30 times the plate thickness. SPO results compare well when an inert VS 1.6 anti-tank mine is buried in a large concrete soil box. Nonlinear hysteresis models of the soil alone and for the soil-plate interaction are useful in understanding these results, since off-the mine vs. on-the mine tuning curves shapes behave significantly different, indicating some false alarms can be eliminated.

11:20

3aSA10. Characterization of air-coupled ultrasonic receivers for nonlinear Rayleigh wave nondestructive evaluation. David E. Torello (Mech. Eng., Georgia Inst. of Technol., 790 Atlantic Dr., Atlanta, GA 30332, david.torello@gmail.com), Jin-Yeon Kim (Civil and Environ. Eng., Georgia Inst. of Technol., Atlanta, GA), Jianmin Qu (Civil and Environ. Eng., Northwestern Univ., Evanston, IL), and Laurence J. Jacobs (Civil and Environ. Eng., Georgia Inst. of Technol., Atlanta, GA)

Nonlinear Rayleigh wave measurements can be used to determine damage precursors and measure microstructural material changes while only requiring one-sided access to a specimen. These measurements are extremely sensitive to coupling conditions between the specimen surface and the generating and receiving transducers used in propagation and measurement of the Rayleigh waves. Air-coupled detection offers many advantages to traditional contact techniques because it mitigates these experimental coupling concerns with a reasonable cost as contrasted with other non-contact methods. However, to use these devices to measure absolute nonlinearity in a specimen requires a detailed understanding of the transducers and the experimental setup. This work provides a combined numerical and analytical approach to understanding the received signal and uses this information to determine a framework for calculation of absolute measures of material nonlinearity.

Session 3aSC

Speech Communication: Various Topics in Speech Communication

Melissa M. Baese-Berk, Chair
Michigan State University, Oyer Center B-7, East Lansing, MI 48824

Contributed Papers

9:00

3aSC1. Human spoken language diversity and the acoustic adaptation hypothesis. Ian Maddieson (Dept. of Linguist, UNM, Univ. of New Mexico, MSC03-2130, Albuquerque, NM 87131-0001, ianm@berkeley.edu) and Christophe Coupé (Laboratoire Dynamique de Langage-CNRS, Lyon, France)

Bioacousticians have argued that ecological feedback mechanisms contribute to shaping the acoustic signals of a variety of species and anthropogenic changes in soundscapes have been shown to generate modifications to the spectral envelope of bird songs. Several studies posit that part of the variation in sound structure across spoken human languages could likewise reflect adaptation to the local ecological conditions of their use. Specifically, environments in which higher frequencies are less faithfully transmitted (such as denser vegetation or higher ambient temperatures) may favor greater use of sounds characterized by lower frequencies. Such languages are viewed as "more sonorous." This paper presents a variety of tests of this hypothesis. Data on segment inventories and syllable structure is taken from LAPSyD, a database on phonological patterns of a large worldwide sample of languages. Correlations are examined with measures of temperature, precipitation, vegetation, and geomorphology reflecting the mean values for the area in which each language is traditionally spoken. Major world languages, typically spoken across a range of environments, are excluded. Several comparisons show a correlation between ecological factors and the ratio of sonorant to obstruent segments in the languages examined offering support for the idea that acoustic adaptation applies to human languages.

9:15

3aSC2. Prosodic features of stance acts. Valerie Freeman (Univ. of Washington, Box 352425, Seattle, WA 98195, valerief@uw.edu)

While textual aspects of stance (attitudes/opinions) have been well studied in conversation analysis and computational models, acoustic-phonetic properties have received less attention. Recent work (2014 and 2015) has found that variations in prosodic measures (speech rate, vowel duration, pitch, and intensity) are correlated with stance presence and strength in unscripted speech, and stances with different discourse functions may be distinguishable by the shapes of their pitch and intensity contours. Building on these early findings, this presentation investigates prosodic properties of various stance-act types in spontaneous conversation (e.g., opinion-offering and soliciting, (dis)agreement, persuasion, rapport-building). The dataset contains over 32,000 stressed vowels from content words spoken by 40 speakers drawn from an audio corpus of dyads engaged in collaborative tasks. Speaker-normalized vowel duration, pitch, and intensity are automatically extracted from time-aligned transcriptions that have been hand-annotated for stance strength, polarity, and act type. Results show that changes in the prosodic measures combine to distinguish several notable stance-act types, including: weak-positive agreement, rapport-building agreement, reluctance to accept a stance, stance-softening, and backchanneling. Pitch and intensity contours over vowel duration are particularly illustrative, suggesting a future avenue in examining contours over whole stance acts.

9:30

3aSC3. Acoustic cues to the [j]-[i] distinction in American English. Zachary Jaggers (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, zackjaggers@nyu.edu)

Existing lexical items suggest that American English exhibits a [j]-[i] distinction (e.g., *pneumonia* [numonjə], *Estonia* [ɛstoniə]). This study tests if such a distinction can be experimentally elicited in both existing and new items and what acoustic cues most consistently convey it. A sentence reading task elicits the distinction by native speakers of American English using orthographically paired nonce names: 'y' stimuli (e.g., *Chobya*) expecting [jV] productions, 'i' stimuli (e.g., *Shabia*) expecting [iV] productions. Stimuli are controlled and diversified along the factors of place and manner of the preceding consonant and word position (initial vs. medial). Multiple acoustic factors of [$_{V]}$ sequences are measured and tested against each other as predictors of stimulus orthography, thus as cues to any elicited distinction, in a generalized linear mixed-effects model. Productions of 'y' stimuli are predicted by significantly earlier transition to the following vowel (represented by timepoint of the F2 maximum), lower F1, and lower intensity. This confirms the presence of the distinction and supports a constriction/height-based classification (Padgett 2008). A significant difference in F2 is not observed; these results are therefore not consistent with a classification of [j] as a coronal sound and [i] as dorsal (Levi 2008).

9:45

3aSC4. Does knowledge of probabilistic pronunciation patterns aid perception? Shinae Kang, Clara Cohen (Linguist, Univ. of California, Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, cpccohen@berkeley.edu), and Rozina Fonyo (San Jose State Univ., Berkeley, CA)

Words which are probable in their morphological paradigms tend to have lengthened affixes (Cohen, 2014; Kuperman *et al*., 2007). Here, we ask whether listeners use the pattern to aid perception. If so, paradigmatically probable words with lengthened affixes should be perceived more quickly than similarly lengthened improbable words. In two experiments (Experiment 1: phoneme monitoring; Experiment 2: lexical decision), we measured listeners' reaction time (RT) to 50 English verbs with [-s] suffixes (e.g., *looks*, *breaks*). Each suffix was adjusted in duration to either a normalized proportion of stem duration, shortened by 25% of the normalized duration, or lengthened by 25%. In Experiment 2, paradigmatic probability did not affect RT, but short words had slower RTs than normalized or lengthened words, suggesting that generally reduced suffix duration impedes perception. In Experiment 1, RT decreased with increased probability for the long condition, but not for the normalized condition. This suggests that a match between suffix length and paradigmatic probability facilitates perception. However, RT also decreased in the short condition, suggesting that when the stimulus was most difficult to perceive, listeners drew on general probabilistic information to aid perception. Our results further indicate that the effect of paradigmatic probability on perception is task-dependent.

10:00–10:15 Break

10:15

3aSC5. Extracting accent information in noise-vocoded speech in Japanese. Yukiko Sugiyama (Keio Univ., Hiyoshi 4-1-1, Kohoku-ku, Yokohama 223-8521, Japan, yukiko_sugiyama@mac.com)

This paper gives a preliminary report of a study that examined the perception of Japanese accent in noise-vocoded speech, where spectral information is substantially compromised while keeping the amplitude envelope intact. While the F0 is known to be the primary cue for accent in Japanese, it is not certain if secondary cues exist. Acoustic analyses conducted in previous studies show mixed results in this regard. The present study attempts to find non-F0 correlates of Japanese accent by conducting a perception study that used noise-vocoded speech. Twelve native speakers of Tokyo Japanese heard ten minimal pairs of final-accented and unaccented words (e.g., /hana*/ "flower" when accented vs. /hana/ "nose" when unaccented) embedded in a carrier sentence. The results obtained so far show that the listeners' accuracy exceeded chance level, suggesting that some acoustic information in the stimuli was present for the listeners to identify words. At the same time, relatively large individual differences were observed in their performance, indicating that some listeners were better at eliciting accent information than others. Implications of the results for the nature of Japanese accent and the perception of accent will be discussed.

10:30

3aSC6. Acoustic and perceptual correlates of subjectively rated sentence clarity in clear and conversational speech. Sarah H. Ferguson and Shae D. Morgan (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu)

Young adults with normal hearing and older adults with hearing loss performed subjective ratings of speech clarity on sentences spoken by all 41 talkers of the Ferguson Clear Speech Database. The sentences were selected from the CID Everyday Sentence lists and were produced under instructions to speak in a conversational manner and in a clear speaking style. A different set of 14 sentences was recorded in each style. Rated clarity will be compared between the two listener groups as well as among subgroups of talkers who differ in demographic and other characteristics. Clarity data will also be analyzed in conjunction with perceptual and acoustic data obtained in other investigations to reveal the relationship between vowel intelligibility and sentence clarity as well as the acoustic features that underlie perceived sentence clarity for different listener groups.

10:45

3aSC7. Developmental trajectory for perception of nonnative-accented sentences. Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

The ability to recognize words under adverse listening conditions slowly develops throughout childhood. For example, children's speech perception in noise or reverberation does not reach maturity until adolescence. Much less is known about the developmental trajectory for children's word recognition under adverse listening conditions stemming from the talker, such as in cases of unfamiliar dialects or accents. To investigate development of word recognition with an unfamiliar accent, 5- to 15-year olds and young adults were presented with native- and Japanese-accented sentences in quiet and noise. Results showed that although 11- to 12-year olds' word recognition for the native in noise condition was similar to adults, the oldest children in the study (i.e., 14- to 15-year olds) did not demonstrate adult-like word recognition for the nonnative talker, with a large performance gap for the noise-added condition. Therefore, the developmental trajectory for word recognition with unfamiliar accents is quite protracted, similar to word recognition in noisy or reverberant environments. Because children's performance for the nonnative talker was depressed relative to adults even in quiet, children's difficulty perceiving unfamiliar accents is likely a consequence of cognitive-linguistic developmental factors or insufficient linguistic experience rather than sensory factors.

11:00

3aSC8. A comparison of speech enhancement methods to extract Lombard speech in an external noise field. Ghazaleh Vaziri, Christian Giguère (Rehabilitation Sci., Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H 8M5, Canada, gvazi024@uottawa.ca), Hilmi Dajani (Elec. Eng. and Comput. Sci., Univ. of Ottawa, Ottawa, ON, Canada), and Nicolas Ellaham (Rehabilitation Sci., Univ. of Ottawa, Ottawa, ON, Canada)

The tendency of talkers to increase their vocal effort in noise, known as the Lombard effect, depends on many factors including the type and level of noise. To investigate the influence of these different factors while wearing hearing protectors, the Lombard effect needs to be elicited by an external noise field (e.g., loudspeakers) rather than the traditional method (e.g., headphones). While the Lombard speech produced may be contaminated by the eliciting noise, the alterations in the talker's voice are more realistically accounted for by such methodology. The problem of recovering sound-field elicited Lombard speech has not been studied extensively. In this study, two noise suppression techniques, direct waveform subtraction and adaptive filtering, are used to recover the Lombard speech produced in simulated conditions using a manikin. To assess the performance of the two methods, the size of noise reduction is compared and basic speech characteristics (e.g., pitch and energy) and objective quality and intelligibility measures (e.g., SII and STI) are extracted from the clean and enhanced Lombard speech. Preliminary results show that the simulated Lombard speech could be accurately recovered, which is useful to extend to speech production in noise with real talkers.

Session 3aSP

Signal Processing in Acoustics: Random Matrix Theory in Acoustics and Signal Processing

James Preisig, Chair
JPAnalytics LLC, 638 Brick Kiln Road, Falmouth, MA 02540

Invited Papers

9:00

3aSP1. Application of random matrix theory to acoustic modeling and signal processing. Kathleen E. Wage (George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, kwage@gmu.edu)

Random matrix theory (RMT) characterizes the eigenvalues and eigenvectors of matrices composed of random entries [Bai/Silverstein, Springer, 2010]. Random matrices play an important role in a variety of acoustics and signal processing applications. For example, mode propagation through internal waves can modeled using random scattering matrices Hegewisch and Tomsovic showed that mode scattering can be analyzed using RMT [JASA, 2013]. In signal processing, the sample covariance matrix is an example of a random matrix that can be analyzed using RMT techniques. RMT differs from classical statistics because it derives results in the limit as both array size and number of measurements approach infinity. This type of asymptotic analysis is crucial because it facilitates the investigation of large dimensional matrices in scenarios with limited numbers of measurements. While results are derived in the infinite limit, many authors have shown that RMT provides useful predictions for finite size matrices derived from finite numbers of measurements. Research in RMT has grown substantially since the 1950s, and new theoretical results are rapidly emerging. This talk reviews several key results from the RMT literature and provides examples to illustrate the analysis of signal processing algorithms and acoustic propagation using these powerful methods.

9:30

3aSP2. Random matrix theory enabled performance analysis and algorithms for underwater signal processing. Raj R. Nadakuditi (Univ. of Michigan, 1012 Pontiac Trail, Unit 5, Ann Arbor, MI 48105, rajnrao@umich.edu)

Random matrices arise naturally in many undersea signal processing applications such as sonar and underwater acoustic communications. For example, the matrix formed by stacking a noisy time series of observations collected at a sensor array alongside each other is a random matrix. Random matrix theory provides a mathematical framework for reasoning about and understanding the structure in such noisy matrix-valued signals in an analogous manner to how Fourier analysis provides us a mathematical framework for reasoning about and understanding the structure in noisy vector valued signals. We highlight some recent breakthroughs in random matrix theory that have allowed us to predict the fundamental performance limits of weak signal detection, estimation and classification and discuss some recent successes where the theory has led to the development of powerful new algorithms for better estimating weaker signals than previously thought possible.

Contributed Papers

10:00

3aSP3. Probability distribution of multiple-target data snapshots applied to large-aperture array processing. Jorge E. Quijano (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd. (Ring Road), Victoria, BC V8P 5C2, Canada, jorgeq@uvic.ca) and Lisa M. Zurk (Elec. and Comput. Eng. Dept., Portland State Univ., Portland, OR)

We consider sonar array target detection and azimuth estimation in experimental scenarios consisting of large-aperture horizontal line arrays of hydrophones in the water column. In such scenarios, successful distinction of quiet targets from the background noise strongly depends on having sufficient independent data snapshots available for computation of adaptive beamforming weights. In this work, the discrete Fourier transform matrix is used for projecting the array data into a sparse domain, from which the theoretical probability density function of each (sparse-domain) datum can be obtained. It is shown that the sparse data exhibits a scaled chi-squared distributed magnitude, with a scale factor that depends on signal-to-background noise ratio and array size. Using this theoretical distribution and Monte Carlo simulations, minimum experimental requirements for target detection are quantified and highlight the tradeoff between data sample size, array aperture, and signal-to-noise ratio. In addition, a statistical test to reduce the number of false detections at the output of a beamformer is proposed and applied to simulated and experimental data from the Shallow Water Array Performance experiment.

10:15

3aSP4. The surprising sample covariance matrix: Unexpected characteristics and understanding them. Atulya Yellepeddi (Elec. Engineering/Appl. Ocean Phys. and Eng., MIT/WHOI, 77 Massachusetts Ave., Bldg. 36-683, Cambridge, MA 02139, atulya@mit.edu) and James Preisig (JPAnalytics, LLC, Falmouth, MA)

The Sample Covariance Matrix (SCM) is important to many problems in acoustic signal processing in both time-domain and frequency-domain processing. However, some careful analysis suggests that the SCM contains

a variety of surprises. For instance, not all elements of the SCM have the same error—the error of a particular element depends on the location of the element in the matrix and is different for SCMs of real-valued processes and complex-valued processes. Moreover, when the samples of the process used to compute the SCM are correlated, the sample covariance matrix behaves differently yet again. In particular, for frequency-domain sample covariance matrices, which are common in underwater acoustic communication and sonar signal processing, the SCM obtained using a tapped delay line is a better estimate of the covariance matrix than that obtained using independent samples—a most counterintuitive result. In this talk, we present a unified analysis technique using ergodic theory that predicts a variety of unexpected characteristics of the SCM, and explain the reasons behind some of them. Given the wide variety of algorithms that rely on the SCM, understanding this behavior is crucial to designing robust and accurate algorithms.

WEDNESDAY MORNING, 4 NOVEMBER 2015 RIVER TERRACE 2, 7:45 A.M. TO 10:40 A.M.

Session 3aUW

Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Animal Bioacoustics: 50 Years of Underwater Acoustics under ASA

David L. Bradley, Cochair
Penn State University, PO Box 30, State College, PA 16870

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

Chair's Introduction—7:45

Invited Papers

7:50

3aUW1. Underwater Acoustics in 1950 and 1972. David L. Bradley (Penn State Univ., PO Box 30, State College, PA 16870, dlb25@psu.edu)

The National Academy (National Research Council) published "A Survey Report on Basic Problems of Underwater Acoustics Research" in 1950 and the Acoustical Society of America, in 1972, published a series of papers that were a 20-year review of various aspects of underwater acoustics. Between the two references, they provide a background of accomplishments and the perspective of scientists at that time, of the major issues in underwater acoustics. This paper is a summary of that information, and provides an "introduction" to the next 5 decades of research that is the focus of this special session. Though not the intent, a comparison of the research accomplished in later years with the proposed effort will provide a basis for further discussion.

8:10

3aUW2. Project MiMi—Miami-Michigan—The discovery of phase coherent acoustic transmission. Harry A. DeFerrari (Ocean Sci., Univ. of Miami, 1148 N E 89 St., Miami, FL 33138, hdeferrri@rsmas.miami.edu)

Project MIMI aimed to observe the temporal fluctuations of low frequency CW acoustics transmission. At the time, the conventional wisdom was that CW phase was randomized after transmission through the ocean. Experiments (transmission between ships) verified that finding and ray theoretical considerations confirmed that uncertainty in the depth of the upper turning point resulted in large variations in travel time—hence random phase. But, some classified work suggested that the ocean transmission was perhaps stable and coherent. John Stienberg experimental group at Miami installed a 400 Hz. bottom mounted directional source aimed at Bimini, Bahamas, but could barely detect the signal. Ted Birdsall provided signal processing gain (36 dB) with the Phase Coherent Demodulator (PCD) that lead to the discovery of phase coherence. The Miami—Michigan (MiMi) team resulted. John Stienberg found that phase variations had periodicities corresponding to surface wave, internal wave, tide and lunar periodicities. He concluded that by "Oceanographic Acoustics" one could make acoustic measurements of oceanographic variations. Birdsall introduces M-sequences ("Gain of CW with the resolution of a pulse") to observe pulse arrivals This approach served as a template for dozens of future fixed-system experiments by the basic research community up until the present time.

8:30

3aUW3. Sixty years studying wave propagation in random media at the Applied Physics Laboratory. Terry E. Ewart and Daniel Rouseff (Appl. Phys. Lab., Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105, teewart@gmail.com)

Ocean acoustics has been a useful avenue for testing evolving theories for Wave Propagation in Random Media (WPRM). These theories generally assume that the index of refraction statistics are stable in space and time, an assumption proven reasonably true in the deep ocean for acoustic paths away from boundaries. In the present work, results from 60 years of theoretical and experimental WPRM research at the University of Washington's Applied Physics Laboratory (APL) are reviewed. The first experiment was performed in 1959 to test theories for amplitude fluctuations based on the Born approximation. The Rytov approximation (from Russian literature) for calculating the log-amplitude fluctuations was also evaluated. Conclusion: neither applied. Experiments in 1971 and 1977 measured acoustic fluctuation statistics for an 18 km acoustic path at sonar-relevant frequencies, 2–13 kHz. A 1985 experiment under Arctic ice used 2–16 kHz signals over a 6 km path. These experiments are discussed together with theoretical issues based on the Moment Equation method to provide one viewpoint on the history of ocean acoustic WPRM. The following translation of Voltaire is appropriate: "The ancients when reasoning about physics without the enlightenment of experiments are like blind men explaining the nature of colors to other blind men."

8:50

3aUW4. ONR arctic acoustics 1978—1998. Arthur B. Baggeroer (Mech. and Elec. Eng., Massachusetts Inst. of Technol., Rm. 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu) and Peter N. Mikhalevsky (Leidos Corp, Arlington, VA)

The Arctic Program Office of the Office of Naval Research ten Arctic field programs from 1978–1994 under the visionary leadership of program managers Dr. G. Leonard Johnson and Dr. Tom Curtin. During this period, over ten ice camps in both the western Arctic (Beaufort Sea) and the eastern Arctic (Nansen and Pole Abyssal Plains were manned and four ice breakers served as platforms in the marginal ice zone (Fram Straits). Since the cost of the support logistics for Arctic field programs is so very high, these experiments were multidisciplinary and almost all had an acoustic component. Some of the highlights were transoceanic reverberation, seismic reflection and refraction, random channels for time and Doppler spreading, target detection, matched field processing,ocean acoustic tomography, seismicity, and ambient noise were among the many topics examined. There were also robust efforts advancing data acquisition. Large, two dimensional horizontal arrays with both cabled and "WIFI" telemetry, large vertical arrays, precision sensor navigation, and sophisticated remote instrumentation buoys were deployed. With the end of the "Cold War" the last field program was in 1994 and the ONR Arctic program eventually was disestablished. Now, the retreat Arctic ice cover and Arctic Ocean warming has reinvigorated ONR's interest in the Arctic and after two decades ONR field programs are planned for the near future.

9:10–9:25 Break

9:25

3aUW5. Nonlinear acoustics and the Acoustical Society of America. Thomas G. Muir (The University of Texas at Austin, Appl. Res. Labs., P.O. Box 8029, Austin, TX 78713, tgmuir@earthlink.net), David G. Browning (None, Kingston, RI), and Kenneth G. Foote (Woods Hole Oceanographic Inst., Woods Hole, MA)

The modern era of nonlinear underwater acoustics had its roots in a remarkable session at the 57th ASA meeting at Providence RI in June, 1960, chaired by Westervelt, of Brown University, who also presented his remarkable new theory of the parametric acoustic array. This conceptual "device" enables highly directive low frequency sound to be created from the nonlinear interaction of two high frequency radiations of high directivity, which was confirmed in a model- tank study paper in this session by Brown Univ. Professor Robert Beyer and student Bellin. JASA has been the major worldwide forum for publications in nonlinear underwater acoustics, ever since, with much of the U.S. work done at the Navy Underwater Sound Laboratory and its successor, the Naval Underwater Systems Center, as well as the Applied Research Laboratories of the University of Texas at Austin. Topics have included parametric arrays, the nonlinear generation of harmonic radiations and transients, acoustic saturation, nonlinear transmitting and receiving sonar and imaging, and many practical applications to underwater sound and acoustical oceanography. Current and/or developing applications include underwater communications, backscattering measurements, penetration of narrow beams into sediments, sub-bottom profiling, and fish school quantification and classification. These and other topics are reviewed and recent implementations are described. [Work supported by ARL:UT Austin.]

9:45

3aUW6. Deep-water ocean acoustic propagation: Observations. Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, pworcester@ucsd.edu)

There is a rich history of long-range, low-frequency, deep-water ocean acoustic propagation measurements extending back to the discovery of the deep sound channel by Ewing and Worzel in 1944. Experiments up to the 1970s focused on measuring parameters in the sonar equation, including transmission loss and ambient noise, motivated in part by the development of the U.S. Navy Sound Surveillance System (SOSUS). These measurements generally used wideband explosive sources or narrowband transducers. Beginning in the 1970s, low-frequency, broadband transducers and vertical receiving arrays that could operate autonomously and be moored for extended periods were developed in connection with the new field of ocean acoustic tomography. These developments allowed individual multipaths and modes to be resolved and long time series collected, so that the fluctuations in the received signals could be quantified. At about the same time, advances in characterizing ocean internal wave and mesoscale variability provided information on the causes of the acoustic fluctuations that could be used in theoretical calculations. The study of deep-water propagation and the development of ocean acoustic tomography have been intertwined ever since. The applications of deep-water propagation now extend beyond military uses to ocean acoustic remote sensing (active and passive), communication, and navigation.

10:05

3aUW7. Shallow water experiments and results. James Lynch, Timothy F. Duda, Arthur E. Newhall, and Ying T. Lin (Woods Hole Oceanographic, MS # 11, Bigelow 203, Woods Hole Oceanographic, Woods Hole, MA 02543, jlynch@whoi.edu)

Shallow water acoustics has flourished over the past fifty years, but most especially over the last 25 years. After World War II, the Cold War between the USSR and the West focused the emphasis in ocean acoustics on deep, "blue water." After the Cold War waned in 1990, the emphasis changed to "brown water" coastal acoustics studies. In this paper, we will look at the advances over the past half-century in shallow water acoustics, with emphasis on the experimental results from acoustics, oceanography, marine geology, and marine biology.

Contributed Paper

10:25

3aUW8. Bottlenose dolphins direct sonar clicks off-axis of targets to maximize Fisher Information about target bearing. Laura Kloepper (Dept. of Biology, Saint Mary's College, 185 Meeting St. Box GL-N, Brown Univ.Dept. of Neurosci, Providence, RI 02912, laurakloepper@gmail.com), Yang Liu, and John R. Buck (Univ. of Massachusetts Dartmouth, Dartmouth, MA)

Aiming the sonar beam directly at a potential target maximizes the detection information available to an echolocating animal. In contrast, estimating the angular location of a target relies on changes in the received echo spectrum due to the frequency dependence of the transmit beam pattern. The Fisher Information quantifies the available information on angular location in terms of the sensitivity of the received echo spectrum as a function of bearing. We calculated the Fisher Information for a dolphin's echolocation signal and determined the maximum angular location information occurs when the sonar beam is such that the target falls slightly off-axis. To compare the predicted beam aim to the actual beam aim of an echolocating dolphin, we recorded the echolocation signals of a bottlenose dolphin with a 16-element hydrophone array while the animal performed a target detection task. The dolphin consistently pointed its sonar beam 7 degrees away from the target, which is the beam aim that maximizes the Fisher Information for the dolphin signal. [Work supported by NSF and ONR.]

WEDNESDAY MORNING, 4 NOVEMBER 2015 GRAND BALLROOM FOYER, 9:00 A.M. TO 12:00 NOON

Session

Exhibit and Exhibit Opening Reception

The instrument and equipment exhibit is located near the registration area in the Grand Ballroom Foyer.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

Exhibit hours are Monday, 2 November, 5:30 p.m. to 7:00 p.m., Tuesday, 3 November, 9:00 a.m. to 5:00 p.m., and Wednesday, 4 November, 9:00 a.m. to 12:00 noon.

Coffee breaks on Tuesday and Wednesday mornings (9:45 a.m. to 10:30 a.m.) will be held in the exhibit area as well as an afternoon break on Tuesday (2:45 p.m. to 3:30 p.m.).

The following companies have registered to participate in the exhibit at the time of this publication:

Brüel & Kjær Sound & Vibration Measurement—www.bksv.com

Freudenberg Performance Materials—www.Freudenberg-pm.com

G.R.A.S Sound & Vibration—www.gras.us

PCB Piezotronics—www.pcb.com/

Sensidyne—www.sensidyne.com

Springer—www.Springer.com

Teledyne Reson—www.teledyne-reson.com

Session 3pAA

Architectural Acoustics: AIA CEU Course Presenters Training Session

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

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

All are welcomed, but TCAA membership and sign-in/sign-out attendance for the entire two hours of this workshop are required to qualify as an authorized presenter for this AIA/CES short course.

Chair's Introduction—1:00

Invited Papers

1:05

3pAA1. TCAA short course presentation material. K. A. Hoover (McKay Conant Hoover, Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects, called "Architectural Acoustics." An architect can earn one continuing education unit (CEU) by attending this short course, if it is presented by a qualified member of TCAA. The course covers topics in sound isolation, mechanical system noise control, and finish treatments. This paper will cover the course material in order to prepare and qualify potential presenters. In order to qualify as an authorized presenter for this AIA/CES short course, attendance at this workshop and membership in TCAA are required.

2:05

3pAA2. AIA/CES Provider registration and reporting requirements. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects. The TCAA short course is called "Architectural Acoustics" and attendance at this one-hour long course can earn an architect one continuing education unit (CEU) with HSW Credit (Health Safety and Welfare). This paper will cover the administrative requirements of the AIA/CES, to prepare potential presenters. These requirements include the proper handling of paperwork, so that AIA members may receive credit for the course. Also, the manner in which the course is given is dictated by AIA requirements. TCAA membership and attendance at this workshop are required to qualify as an authorized presenter for this AIA/CES short course.

Session 3pBA

Biomedical Acoustics: Therapeutic Ultrasound, Microbubbles, and Bioeffects II

Wayne Kreider, Chair
CIMU, Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Contributed Papers

1:00

3pBA1. *In vivo* biodistribution of fluorescently tagged magnetic microbubbles for cavitation enhancement with real time passive acoustic mapping. Calum Crake, Robert Carlisle, Joshua Owen, Sean Smart, Christian Coviello, Constantin Coussios, and Eleanor P. Stride (Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

Previous work has demonstrated the potential of magnetically functionalized microbubbles to localize and enhance cavitation activity under focused ultrasound exposure *in vitro*. The aim of this study was investigate magnetic targeting of microbubbles for promotion of cavitation *in vivo*. Fluorescently labeled magnetic microbubbles were intravenously injected into a murine xenograft model. Cavitation was induced using a 0.5 MHz single element focused transducer at peak negative focal pressures of 0.1–1.0 MPa and monitored in real-time using simultaneous B-mode imaging and passive acoustic mapping. Magnetic targeting was found to increase the amplitude of the cavitation signal as compared with untargeted bubbles and a commercial ultrasound contrast agent. Post exposure magnetic resonance imaging indicated a correlation between cavitation activity and deposition of magnetic nanoparticles in the tumor volume. Magnetic targeting was similarly associated with an increase in the fluorescence intensity measured from the tumors following the experiments; the highest levels corresponding to the maximum cavitation signals. These results indicate a strong correlation between cavitation activity and increased uptake of both molecular species and nanoparticles in tumors; and that the effect can be enhanced by magnetic targeting.

1:15

3pBA2. Acoustic methods for modulating the cavitation initiation pressure threshold. Hedieh A. Tamaddoni, Alexander P. Duryea, and Timothy L. Hall (Univ. of Michigan, Biomedical Eng., 2109 Gerstacker, Ann Arbor, MI 48109, alavi@umich.edu)

The objective of this study is to develop tissue protection methods by modulating bubble cloud initiation pressure thresholds. Specifically, we investigated pressure thresholds to initiate cavitation bubble clouds by the shock scattering mechanism. Cavitation initiation displays a stochastic nature affected by existing nuclei populations in the medium. We hypothesized that by applying proper low pressure pulse sequences before and/or during histotripsy therapy, initiation pressure threshold and growth of cavitation bubble cloud could be modified. We applied histotripsy and cavitation suppressing pulses in both water and agarose gel for pulse repetition rates of 1, 10, and 100 Hz. Acoustic backscatter signals and optical imaging were used to detect and monitor initiation, maintenance, and growth of resulting cavitation bubble cloud. Results demonstrated that the use of cavitation suppressing pulses can increase the cavitation threshold by 20% in the targeted space. Furthermore, we showed these acoustic sequences could modify the shape and density of the bubble cloud. By applying the cavitation suppressing pulses, we were able to generate a dense cavitation bubble cloud in the focus while decreasing scattered cavitation in the peripheral zone.

1:30

3pBA3. Experimental and numerical evaluation of the effect of stone size on fracture by burst wave lithotripsy. Madeline J. Hubbard, Barbrina Dunmire (Ctr. for Medical and Industrial Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Wayne Krieder, Michael R. Bailey (Ctr. for Medical and Industrial Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, 1013 NE 40th St, Seattle, WA 98105, amax38@u.washington.edu)

Burst wave lithotripsy (BWL) is an experimental treatment to noninvasively fragment kidney stones using bursts of focused ultrasound. Preliminary simulations with a linear elastic model showed that resonance creates concentrated stresses, which may help predict locations of fractures in the stones. In this study, we aimed to demonstrate this correlation by comparing simulations to experimental data. Cylindrical stones of variable size (4–14 mm diameter, 10 mm length) made from BegoStone plaster were treated in a water bath for 10 min using a 170 kHz focused transducer at a focal pressure of 6.5 MPa. Locations of first fractures in the stones correlated well with the location of peak stress predicted in the linear elastic model. Simulated peak surface stress in the stones decreased as stone diameter increased, with the exception of a spike near $d = 12$ mm, which matches the half longitudinal wavelength in the stone. Experimentally, a corresponding overall increase in time to first fracture was observed with diameter, except for a drop at $d = 12$ mm. The results are encouraging that the model may help direct further optimization of BWL. [Work supported by NIH through DK043881, DK104854, EB007643, and NSBRI through NASA NCC 9-58.]

1:45

3pBA4. Fast passive cavitation mapping with angular spectrum approach. Costas Arvanitis, Nathan McDannold (Radiology, Harvard Med. School, Brigham and Women's Hospital, 221 Longwood, Rm. 514A, Boston, MA 02115, arvanitis.costas@gmail.com), and Gregory Clement (Biomedical Eng., Cleveland Clinic Lerner Res. Inst., Cleveland, OH)

Several emerging focused ultrasound (FUS) therapies harness the effects produced by acoustic cavitation. Passive acoustic mapping (PAM) can be used to characterize and visualize the microbubble oscillations and guide these procedures. Here, we propose and demonstrate that angular spectrum approach (ASA) can be used to back-project the passively recorded, by an array of receivers, diverging pressure waves generated from oscillating microbubbles to perform rapid PAM reconstructions. In the present work, acoustic cavitation is studied transcranially in non-human primates using an integrated ultrasound and MRI-guided clinical FUS system. In addition, from CT datasets, we also extract the skull acoustic properties and use them as inputs to numerical simulations. Using both the simulated and experimental data we validate the use of ASA to perform PAM reconstructions and compare its performance to time domain PAM. The experimental data demonstrate that ASA can be used to reconstruct frequency-selective PAM. Numerical simulations suggest that ASA reconstructions have the same resolution with time domain PAM, while the reconstruction time was 72 times faster for the same image dimensions. These results suggest that ASA-based PAM reconstructions can provide real-time passive cavitation mapping and, by extension, control over FUS procedures.

2:00

3pBA5. Acoustic levitation device for probing biological cells with high-frequency ultrasound. Brian D. Patchett (Phys., Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, brian.d.patchett@gmail.com), Natalie C. Sullivan (Chemistry, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

High-frequency ultrasound has been shown to be sensitive to changes in the cell cytoskeletal makeup during a variety of biological processes. It can therefore be used to study cellular responses such as T-cell activation and cancer metastasis. To date, obtaining cytoskeletal properties with high-frequency ultrasound has required the use of monolayer cell cultures. Ultrasonic signals from these cultures can be problematic due to the interference of reflections from the culture-plate well. The cell structure is also often deformed from its native state due to adhesion to the culture plate. The objective of this study was to develop an acoustic levitation device that creates a monolayer of cells suspended in a fluid in order to simulate their environment *in vivo*. By using a 250 kHz transducer, standing waves were generated in a suspension of polyethylene microspheres (53–63μm diameter) in distilled H_2O. Microspheres formed layers at the standing-wave nodes. Varying the shape of the transducer voltage waveform had a significant effect on layer thickness, with a square waveform creating thinner, more distinct layers than a sine waveform due to steeper pressure gradients at the nodes. Future work will entail creating a monolayer suspension of biological cells that can be probed with high-frequency ultrasound.

2:15

3pBA6. *In vivo* cavitation thresholds and injury observations related to burst wave lithotripsy. Wayne Kreider (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 School of Medicine, Seattle, WA), Bryan W. Cunitz, Yak-Nam Wang (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Donghoon Lee (Dept. of Radiology, Univ. of Washington School of Medicine, Seattle, WA), Mathew D. Sorensen, Jonathan D. Harper (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Oleg A. Sapozhnikov, Vera A. Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), and Michael R. Bailey (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Burst wave lithotripsy (BWL) is a new non-invasive approach for treating kidney stones that uses short bursts of ultrasound at low duty factors. An important difference between BWL and SWL relates to the potential for generating cavitation in the kidney that could cause injury. To study such cavitation behavior, 17 pig kidneys were exposed *in vivo* to BWL treatments of different amplitudes at frequencies of 170 and 335 kHz. Treatments lasted at least 10 min and were monitored in real time using ultrasound imaging (including B-mode and Doppler modalities) to determine threshold pressures for the onset of cavitation. After treatment, eight of the kidneys were perfusion fixed and scanned using MR imaging sequences designed to detect hemorrhagic injury. Results suggest that pressure thresholds exist below which cavitation detected by ultrasound imaging is very unlikely to occur; moreover, MR images indicate no injury or only minimal injury when cavitation was not detected. Observations at 170 kHz show a repeatable threshold pressure near 5 MPa; data at 335 kHz were less consistent but suggest a higher threshold between 6 and 8 MPa. [Funding support by NIH NIDDK P01-DK043881, R01-DK092197, T32-DK007779, K01-DK104854, R01-EB007643, R01-CA188654, and NSBRI through NASA NCC 9-58.]

2:30

3pBA7. Focused ultrasound-enhanced intranasal delivery of brain-derived neurotrophic factor. Hong Chen, Georgiana Z. Yang, Hoheteberhan Getachew, Camilo Acosta, Carlos S. Sánchez (Dept. of Biomedical Eng., Columbia Univ., 630 West 168th St., PS 19-418, New York, NY 10032, hc2666@columbia.edu), and Elisa E. Konofagou (Dept. of Biomedical Eng. and Dept. of Radiology, Columbia Univ., New York, NY)

The therapeutic use of neurotrophic factors in the treatment of central nervous system diseases has been restrained by their low blood-barrier permeability and rapid degradation in the blood. Intranasal (IN) administration is a promising approach for delivering neurotrophic factors directly to the brain, bypassing the BBB. However, IN delivery has low efficiency and does not offer localized delivery to specific brain sites. The objective of this study was to demonstrate the feasibility of FUS-enhanced IN delivery of brain derived neurotrophic factor (BDNF) at a targeted location. BDNF was administered through IN route in wild-type mice ($n = 7$) followed by FUS sonication at the left caudaputamen in the presence of systemically circulating microbubbles. The contralateral right caudaputamen was used as control for IN delivery only. Immunohistochemistry staining was used to assess the distribution of BDNF and the bioactivity of BDNF in activating the downstream signaling. It was found that FUS enhanced the delivery efficiency of IN administered BDNF at the targeted region, where BDNF penetrated deep into the brain tissue instead of confining within the perivascular spaces as in the contralateral control side. Furthermore, the delivered BDNF reached sufficient concentration to activate the downstream signaling pathway.

2:45

3pBA8. Ultrasound-mediated drug release from nanoscale liposomes using nanoscale cavitation nuclei. Susan Graham, James Kwan, Rachel Myers, Christian Coviello, Robert Carlisle, and Constantin Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, constantin.coussios@eng.ox.ac.uk)

We have previously presented a liposomal formulation of mean size 140 nm, manufactured using DSPE, cholesterol, DSPC and DSPE-PEG at ratios of 65:25:3:7, which exclusively releases encapsulated doxorubicin in the presence of inertial cavitation nucleated by microbubbles (SonoVue®, Bracco) at peak rarefactional pressures in excess of 1.2 MPa at 0.5 MHz (Graham *et al.*, J. Controlled Release, 2014). However, the benefits of cavitation-sensitive liposomes small enough to pass through the leaky tumor vasculature can only be fully realized if they can be triggered by cavitation nuclei which are also small enough to extravasate into the tumor mass. In the present work, we demonstrate that liposomal release comparable to that mediated by SonoVue® microbubbles can be achieved using gas-stabilizing polymeric nanocups of mean diameter 400 nm, at peak rarefactional pressure amplitudes in excess of 1.5 MPa at 0.5 MHz or 4 MPa at 1.6 MHz. Mechanistically, we hypothesize that release occurs once a threshold peak shear rate is exceeded in the fluid surrounding the collapsing microbubble, thus exceeding the critical shear stress on the liposomal surface. This is confirmed experimentally by demonstrating a correlation between release and the maximum power of broadband acoustic emissions received by a passive cavitation detector.

Session 3pED

Education in Acoustics: Acoustics Education Prize Lecture

Ning Xiang, Chair
School of Architecture, Rensselaer Polytechnic Institute, Greene Building, 110 8th Street, Troy, NY 12180

Chair's Introduction—2:10

Invited Paper

2:15

3pED1. Propagation of acoustic education in space and time. Yang-Hann Kim (KAIST, Dept. of M.E., Sci. Town, Daejon-shi 305-703, South Korea, yanghann@kaist.ac.kr)

Sound is every where and any time. Human is always surrounded by sound. However, understanding sound is not always available to everyone. It's concepts start to be introduced from elementary school to college, but implementing sound for his/her purpose is always problematic. One reason why sound is so difficult to understand is because it is not visible. There have bee numerous attempts to make sound visible: using water, many candle lights, string, small particles, and many microphones. Another hurdle that we have to go over to really enjoy and use sound is to have a convenient tool for manipulating sound. However making one's own speaker cannot be tackled by using a home took box. However, we are about to make this dream come true: many good speakers can be found in relatively cheap price and tangible manipulation tools begin to be available. This paper introduce how these tools have been developed and can be used for education. Another issue associated with learning sound is how to make acoustic education available to anyone and anytime: distance education, using Internet can be a solution. Past experience will be introduced: college lecture, short course for industry, Internet lecture, and finally MOOC lecture for acoustics are compared. What their characteristics are, and how to use them are to be discussed.

WEDNESDAY AFTERNOON, 4 NOVEMBER 2015 GRAND BALLROOM 2, 1:00 P.M. TO 2:05 P.M.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Tessa Bent, Chair
Department of Speech and Hearing Sciences, Indiana University, 200 S. Jordan Ave., Bloomington, IN 47405

Chair's Introduction—1:00

Invited Papers

1:05

3pID1. Hot topics in underwater acoustics: Recent advances in underwater unexploded ordnance classification. Aubrey L. Espana, Kevin L. Williams, and Steven G. Kargl (Acoust. Dept., Appl. Phys. Lab. - Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105, aespana@apl.washington.edu)

For decades, broadband sonar has been a key tool in the underwater detection and classification of man-made targets. The DoD's recent focus to clean up discarded WWII-era munitions has pushed a flurry of research and advancements in the areas of target scattering, environmental noise modeling, signal processing, and automatic target recognition (ATR). It is the interdisciplinary nature of this

problem that makes it incredibly challenging, namely, the fact that the ability to classify an object is heavily influenced by the operational choices (grazing angle and vehicle altitude), surrounding environmental factors (target type, burial state, sea state, and bottom topography), and the processing tools (data products, feature-set, and ATR training/testing). This talk will outline the key steps underlying the process of trying to detect and classify underwater unexploded ordnance (UXO). It begins by pushing raw time domain data, which can be generated either experimentally or via models, through a set of signal processing tools to create data products (acoustic color and SAS images). These data products are fed through feature extraction processes, followed by ATR algorithms, in order to arrive at the final classification of an object. Examples to be presented include objects of interest to the remediation of underwater UXO. [Research supported by SERDP and ONR.].

1:25

3pID2. Seismic surveys and marine wildlife: Impacts, the lack thereof, and thoughts on managing both. Douglas P. Nowacek (Nicholas School of the Environment and Pratt School of Eng., Duke Univ. Marine Lab., 135 Duke Marine Lab Rd., Beaufort, NC 28516, dpn3@duke.edu)

Sufficient scientific data exist to conclude that seismic airguns used in geophysical exploration have a low probability of direct harm to most marine life, except at close range where physical injury is a real danger. Further, airguns in some conditions do not appear to disturb animals; however, in other conditions, they result in moderate to extreme behavioral responses and/or acoustic masking over large areas. Additionally, recent studies have reported the presence of seismic survey sound energy over ranges of ~4000 km. While the potential for effects have not even been investigated at such ranges, the presence of the signals must be taken into account when evaluating overall potential for impacts. Mitigation measures have historically focused on reducing immediate harm, but systematically measuring and understanding the full potential for impacts is an important aspect of any responsible development program. The European Union has recognized ocean noise as a pollutant and as an indicator of environmental quality under its Marine Strategy Framework Directive. Given this and the international and transboundary nature of noise from marine seismic surveys, their ubiquity, the presence of numerous other sources of ocean noise, and that incorporating acoustic disturbance into an understanding of population level consequences is progressing, a responsible path forward should focus on the creation of legally binding international commitments and standards for the management and minimization of noise.

1:45

3pID3. Hot topics in "cold" acoustical oceanography. Grant B. Deane (Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St, La Jolla, CA 92093-0238, gdeane@ucsd.edu)

Acoustics is a vital tool for probing the ocean interior, the seafloor, and the sea surface. This talk on Acoustical Oceanography will focus on hot topics in cold places, specifically the Arctic. The rapid loss of sea ice in the arctic circle is leading to an increase in human activities, with implications for oil and gas exploration, fisheries, shipping, and tourism, to name a few areas. Current active and exciting research areas in Acoustical Oceanography include: (1) multipurpose acoustic networks in the Arctic that support the passive monitoring of underwater sound, and remote sensing activities such as acoustic tomography and thermometry, (2) using acoustics to track marine mammals and evaluate their responses to industrial noise, (3) the use of high frequency acoustics to detect and quantify oil layers beneath sea ice, and (4) the use of underwater ambient noise to study glacier dynamics and ice melting in arctic, glacial bay.

Session 3pSC

Speech Communication: Speech Style and Sociophonetics (Poster Session)

Benjamin Munson, Chair
Speech-Language-Hearing Sciences, University of Minnesota, 115 Shevlin Hall, 164 Pillsbury Drive SE, Minneapolis, MN 55455

Authors should be at their posters from 1:45 p.m. to 3:15 p.m. To allow authors an opportunity to see other posters in their session, all posters will be on display from 1:00 p.m. to 3:15 p.m.

Contributed Papers

3pSC1. Schwa reduction and the realization of /p, t/ in casual American English. Ellen Aalders and Mirjam Ernestus (Ctr. for Lang. Studies, Radboud Univ. Nijmegen, Erasmusplein 1, Nijmegen, Gelderland 6525 HT, Netherlands, e.aalders@let.ru.nl)

This study focused on the realization of /p, t/ after schwa (e.g., *support, certificate*) in casual American English. We analyzed the realization of 523 tokens in the Buckeye corpus. Pronunciation variation proved substantial. Some plosives were realized as fricatives or were completely absent; others varied in the duration and voicing of the closure, burst presence, and voice onset time (VOT). Tokens of *suppose*(*d*) showed markedly more reduction than other words: schwa was more often absent (80% versus 28%) and VOT was much shorter (mean 19 ms versus 59 ms). We propose that *suppose(d)* has an additional lexical entry without schwa: *spose(d)*. It is typically assumed that if the schwa is lost, a following voiceless plosive retains the long VOT that is typical in stressed syllable-initial position (e.g., *support* pronounced as [s'p^{h}ort]). We compared VOT in our schwa words, excluding *suppose*(*d*), with that of 405 plosives occurring after /s/ (e.g., *sport*, *still*). Regression analyses showed that VOT was longer in syllable-initial position than after /s/ in consonant clusters (mean 16 ms). Schwa absence did not affect VOT and neither did the duration of schwa when present. This suggests that the reduction processes affecting vowels need not change the realization of consonants.

3pSC2. Influence of clear speech on the word intelligibility of electrolaryngeal speakers. Steven R. Cox and Philip C. Doyle (Health and Rehabilitation Sci., Western Univ., Elborn College Rm. 2200, London, ON N6G1H1, Canada, scox47@uwo.ca)

Electrolaryngeal (EL) speech is often used by individuals who undergo total laryngectomy (Barney, Haworth, & Dunn, 1959; Doyle, 1994). Unfortunately, current EL devices continue to produce an unnatural and mechanical sound quality that can impact speech intelligibility (SI) (Doyle & Eadie, 2005; Hillman, Walsh, Wolf, Fisher, & Hong, 1998; Meltzner & Hillman, 2005). Clear speech (CS) requires speakers to produce speech "as clearly as possible" in an attempt to improve the overall understandability of speech (Picheny, Durlach, & Braida, 1985, 1986) in those with communication disorders (Beukelman, Fager, Ullman, Hanson, & Logemann, 2002; Tjaden, Sussman, & Wilding, 2014). This study addressed alterations in SI through the application of CS by EL speakers. Twelve naïve listeners orthographically transcribed CVC words produced by 10 EL speakers in both habitual and CS modes. Findings indicate that SI improved 1.5% for words spoken using CS (M = 53.25%, *range* = 47 to 64%) compared to words spoken using habitual speech (M = 51.75%, *range* = 41 to 62%). No significant differences in SI were found in either word-initial or word-final phonemes across speaking conditions. Clinical implications of these findings will be discussed.

3pSC3. Judgments of self-identified gay and heterosexual male speakers of American English: Certain phonemes are more salient than others in determining sexual orientation. Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Prior research (Munson, McDonald, DeBoe, & White, 2006; Tracy, Bainter, & Satariano, 2015) has demonstrated that, upon hearing a relatively short utterance, listeners were able to differentiate between self-identified gay and heterosexual male speakers of American English. Furthermore, Tracy *et al.* (2015) discovered that listeners primarily relied on certain vowels (e.g., /æ/, /eɪ/, /ɛ/, /iː/, /oʊ/, /ɑ/, and /uː/) and consonants (e.g., /l/, /n/, and /s/) to determine a speaker's sexual orientation. The present study, based on data collected by Tracy (2015), investigated whether listeners relied on other phonemes when forming their sexual orientation judgments. Listeners were presented with utterances that either included one, two, or three consonants, or one, two, or three vowels. Upon hearing a single phone, such as /ɪ/, /^/, /ɹ/, /θ/, and /j/, listeners were able to differentiate between the speakers. With respect to the consonants, listeners' performance improved as the number of certain consonants (e.g., /l/, /n/, /ɹ/, /θ/, and /j/) in the utterance increased. For example, the utterance /ɹ, j, θ/ resulted in better performance than the utterance /f, m, v/. The results indicate that certain phonemes, more than others, can indicate a speaker's sexual orientation.

3pSC4. Are phonetic contrasts enhanced in clear speaking styles? Outi Tuomainen and Valerie Hazan (Speech, Hearing and Phonetic Sci., UCL, Speech, Hearing and Phonetic Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, o.tuomainen@ucl.ac.uk)

When asked to speak clearly, talkers make adaptations to various acoustic characteristics of their speech. Do these adaptations specifically enhance phonetic contrasts or just result in more global enhancements? For phonetic contrasts, increased discriminability could be achieved by increasing between-category distance, reducing within-category dispersion or both. The LUCID corpus contains 32 iterations per consonant for each of 40 adults for the /s/-/ʃ/ and /p/-/b/ contrasts. Iterations were obtained via picture elicitation in a sentence context in two conditions: when asked to speak casually and clearly. Friction centroids were measured for /s/-/ʃ/ and voice onset times for /p/-/b/. For /s/-/ʃ/, although there was significantly greater distance between centroids in the clear speech condition, within-category dispersion did not differ across speaking styles and there was no significant increase in overall discriminability in the clear condition. For /p/-/b/, in the clear condition, there was a significant increase in VOT distance, a significant decrease in within-category dispersion, resulting in an overall increase in discriminability. These results show that clear speaking styles can lead not only to global enhancements to speaking rate, intensity, pitch characteristics but also to increases in discriminability at the segmental level, although this may be contrast-specific.

3pSC5. Effects of speaking style and context on online word recognition in adverse listening conditions. Suzanne V. van der Feest (Linguist, The Univ. of Texas at Austin, 305 E 23rd St., Austin, TX 78712, suzanne@austin.utexas.edu), Samantha Moses, Amanda Clark (Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX), Danielle Parsons, and Rajka Smiljanic (Linguist, The Univ. of Texas at Austin, Austin, TX)

This study investigated the time-course of word recognition in adverse listening conditions. Specifically, we examined the effect of different listener-oriented speaking styles and semantic context on lexical access in quiet and in noise. Young adult listeners participated in an online visual word recognition experiment. They heard sentences with a high- versus low-predictability semantic context, produced in Conversational (CO), Infant-Directed (IDS), and Clear (CS) speech while fixating two pictures on a screen: a target that matched the last word of the auditory stimulus and a distractor. All sentences were presented either in quiet, or mixed with speech-shaped noise at a −5 dB SNR. Results showed that IDS provided similar perceptual benefits to adult listeners as CS. Relative to low-predictability CO baseline, IDS and CS increased speed of word recognition for high-predictability sentences, in quiet and in noise equally. However, in the quiet condition, lexical access was eventually facilitated by contextual cues even in CO, but listeners in noise reliably focused the target only when a combination of contextual cues and exaggerated acoustic-phonetic cues was available. These findings suggest that both semantic cues and listener-oriented acoustic enhancements are needed for reliable and rapid lexical access, especially in adverse conditions.

3pSC6. Spectral characteristics and formant bandwidths of English vowels produced by American males with different speaking styles. Byunggon Yang (English Education, Pusan National Univ., 30 Changjun-dong Keumjunggu, Pusan 609-735, South Korea, bgyang@pusan.ac.kr)

Speaking styles tend to have an influence on spectral characteristics of produced speech. There are not many studies on the spectral characteristics of speech because of complicated processing of too many spectral data. This study examined spectral characteristics and formant bandwidths of English vowels produced by nine American males with different speaking styles: clear or conversational styles; high- or low-pitched voices. PRAAT was used to collect pitch-corrected long-term averaged spectra and bandwidths of the first two formants of 11 vowels in the speaking styles. Results showed that the spectral characteristics of the vowels varied systematically according to the speaking styles. The clear speech showed higher spectral energy of the vowels than that of the conversational speech while the high-pitched voice did the same over the low-pitched voice. Second, there was no statistically significant difference between B1 and B2 in the speaking styles. B1 was generally lower than B2 reflecting the source spectrum and radiation effect. However, there was a statistically significant difference in B2 between the front and back vowel groups. The author concluded that spectral characteristics reflect speaking styles systematically while bandwidths measured at a few formant frequency points do not reveal style differences properly.

Plenary Session and Awards Ceremony

Christy K. Holland, Chair
President, Acoustical Society of America

Annual Membership Meeting

Presentation of Certificates to New Fellows

Ian C. Bruce – For contributions to models of auditory-nerve fibers

Michele B. Halvorsen – For contributions to the understanding of the effects of sound on fish

Valerie L. Hazan – For contributions to the understanding of the intelligibility of speech

Kirill V. Horoshenkov – For contributions to outdoor sound propagation, remote sensing, and acoustics of porous materials

Wilfried Kausel – For contributions to the understanding of the acoustics of brass instruments

Andone C. Lavery – For contributions to the understanding of ocean microstructure and biology

Jose Sanchez-Dehesa – For contributions to the theory and development of acoustic metamaterials

Introduction of Award Recipient

Carl Wunsch, recipient of the 2015 Walter Munk Award
for Distinguished Research in Oceanography Related to Sound and the Sea
a joint award of The Oceanography Society, the Office of Naval Research, and
the Office of the Oceanographer of the Navy

Presentation of Awards

Kelly Servick, recipient of the Science Writing Award in Acoustics for Journalists for her article
"Eavesdropping on Echosystems" (*Science Magazine*, 21 February 2014)

Trevor Cox, recipient of the Science Writing Award for Professionals in Acoustics for his book
The Sound Book: The Science of the Sonic Wonders of the World (W.W. Norton & Company, 2014)

Yang-Hann Kim, recipient of the 2015 Rossing Prize in Acoustics Education

Allan D. Pierce recipient of the Distinguished Service Citation

Paul D. Schomer recipient of the Distinguished Service Citation

John L. Butler, Silver Medal in Engineering Acoustics

Roy D. Patterson, Silver Medal in Psychological and Physiological Acoustics

Brian G. Ferguson, Silver Medal in Signal Processing in Acoustics

John J. Ohala, Silver Medal in Speech Communication,

OPEN MEETINGS OF TECHNICAL COMMITTEES

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

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

Committees meeting on Tuesday, 3 November

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Orlando
Acoustical Oceanography	7:30 p.m.	River Terrace 2
Animal Bioacoustics	7:30 p.m.	City Terrace 9
Architectural Acoustics	7:30 p.m.	Grand Ballroom 3
Physical Acoustics	7:30 p.m.	St. Johns
Psychological and Physiological Acoustics	7:30 p.m.	Grand Ballroom 7
Structural Acoustics and Vibration	7:30 p.m.	Daytona

Committees meeting on Wednesday, 4 November

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Clearwater
Signal Processing in Acoustics	8:00 p.m.	City Terrace 7

Committees meeting on Thursday, 5 November

Committee	Start Time	Room
Musical Acoustics	7:30 p.m.	Grand Ballroom 1
Noise	7:30 p.m.	Grand Ballroom 2
Underwater Acoustics	7:30 p.m.	River Terrace 2