Session 1aID

Interdisciplinary: Opening Ceremonies, Keynote Lectures

8:00

The Presidents of the European Acoustics Association and the Acoustical Society of America will welcome attendees to Acoustics'17 Boston.

Invited Papers

Keynote Introduction—8:15

8:20

1aID1. Computational analysis of acoustic events in everyday environments. Tuomas Virtanen (Tampere Univ. of Technol., Korkeakoulunkatu 1, Tampere FI-33720, Finland, tuomas.virtanen@tut.fi)

Sounds carry a large amount of information about our everyday environment and physical events that take place in it. Recent advances in machine learning allows automatic methods to analyze this information, for example, by detecting and classifying acoustic events produced by various sources. This allows several applications, for example, in acoustic surveillance, context-aware devices, and multimedia indexing. This talk will present signal processing and machine learning methods that can be used to detect and classify everyday acoustic events originating, e.g., from vehicles, human activity, human and animal vocalizations, in everyday environments. It will describe the scientific challenges in such methods, for example, many sources having highly similar spectral characteristics and multiple sources being active simultaneously. It will explain how state-of-the-art methods based on advanced deep neural network topologies deal with these challenges. The talk will also discuss the practical challenges related to the development of the methods, such as acquisition of data that is used to develop the methods. It will present results from recent evaluations of event detection systems and illustrate them using audio and video examples.

Keynote Introduction—9:15

9:20

1aID2. A sound future for acoustic metamaterials. Steven Cummer (Duke Univ., PO Box 90291, Durham, NC 27708, cummer@duke.edu)

The field of acoustic metamaterials borrowed ideas from electromagnetics and optics to create engineered structures that exhibit desired fluid or fluid-like properties for the propagation of sound. These metamaterials offer the possibility of manipulating and controlling sound waves in ways that are challenging or impossible with conventional materials. Metamaterials with zero, or negative, refractive index for sound offer new possibilities for acoustic imaging and for the control of sound at subwavelength scales. The combination of transformation acoustics theory and highly anisotropic acoustic metamaterials enables precise control over the deformation of sound fields, which can be used, for example, to hide or cloak objects from incident acoustic energy. And active acoustic metamaterials use external control and power to create effective material properties that are fundamentally not possible with passive structures. Challenges remain, including the development of efficient techniques for fabricating large-scale metamaterial structures and, critically, converting exciting laboratory experiments into practically useful devices. In this presentation, I will outline the recent history of the field, describe some of the designs and properties of materials with unusual acoustic parameters, discuss examples of extreme manipulation of sound, and finally, provide a personal perspective on future directions in the field.

10:15–10:40 Break

Session 1aAAa

Architectural Acoustics: Echolocation by People Who are Blind

Monika Rychtarikova, Cochair
Faculty of Architecture, KU Leuven, Hoogstraat 51, Gent 9000, Belgium

David P. Garcia, Cochair
Physics and Astronomy, KU Leuven, Heverlee, Belgium

Chair's Introduction—10:35

Invited Papers

10:40

1aAAa1. Human echolocation in different situations and rooms. Bo N. Schenkman (Speech, Music and Hearing, Royal Inst. of Technol. (KTH), Lindstedtsvägen 24, Stockholm SE-100 44, Sweden, bosch@kth.se)

People, especially when blind, use echolocation to detect obstacles, orient themselves, and get an awareness of their environment. I and coworkers have, with mostly psychophysical methods, studied perceptual aspects of how people accomplish echolocation. Echolocation with long canes while walking was possible but difficult. The effects of the spectral composition of the emitted sounds had no effects. Sound recordings in anechoic and conference rooms from non-walking, static situations, later presented in a laboratory showed a better performance in an ordinary room with reflections, than in an anechoic room. We also found that there was a higher performance with longer sounding sounds than for short clicks. Among the difficulties for the blind are how to avoid masking of sounds. A few blind are exceptionally high performing. An "information-surplus principle" has been proposed. Various information sources are used, but repetition pitch seems more important than loudness for echolocation. Among other sources, timbre may also provide information. There may exist a time gap, acoustic gaze, for how blind people use clicks. It is likely also that there are at least two processes taking place in the hearing system when listening for echoes, one attuned to short sounds and one to long sounds.

11:00

1aAAa2. Auditory recognition of surface texture with various scattering coefficients. Monika Rychtarikova (Faculty of Architecture, KU Leuven, Hoogstraat 51, Gent 9000, Belgium, Monika.Rychtarikova@kuleuven.be), Lukaš Zelem (Faculty of Civil Eng., Dept. of Bldg. Construction, STU Bratislava, Bratislava, Slovakia), Leopold Kritly, David P. Garcia (Phys. and Astronomy, Lab. of Acoust., KU Leuven, Heverlee, Belgium), Vojtech Chmelík (Faculty of Civil Eng., Dept. of Bldg. Construction, STU Bratislava, Bratislava, Slovakia), and Christ Glorieux (Phys. and Astronomy, Lab. of Acoust., KU Leuven, Leuven, Belgium)

Human echolocation is a known ability of people to grasp the information about the surrounding environment from purely acoustic information. However, the extent to what normal sighted and blind people can auditorily recognize the surface texture, such as its roughness or other different sound scattering features, is not completely known. In this paper, we investigate the ability of people to distinguish different types of surfaces by their sound reflections. Reflection patterns from 24 types of surface textures at two different distances were calculated in finite difference method and convolved with "click sound" in order to be used for perception tests. Twenty normally sighted human subjects participated on the listening test experiment.

11:20

1aAAa3. Audible sonar images generated with proprioception for target analysis. Roman B. Kuc (Elec. Eng., Yale, 15 Prospect St., 511 Becton, New Haven, CT 06511, roman.kuc@yale.edu)

Some blind humans have demonstrated the ability to detect and classify objects with echolocation using palatal clicks. An audible-sonar robot mimics human click emissions, binaural hearing, and head movements to extract interaural time and level differences from target echoes. Targets of various complexity are examined by transverse displacements of the sonar and by target pose rotations that model movements performed by the blind. Controlled sonar movements executed by the robot provide data that model proprioception information available to blind humans for examining targets from various aspects. The audible sonar uses this sonar location and orientation information to form two-dimensional target images that are similar to medical diagnostic ultrasound tomograms. Simple targets, such as single round and square posts, produce distinguishable and recognizable images. More complex targets configured with several simple objects generate diffraction effects and multiple reflections that produce image artifacts. The presentation illustrates the capabilities and limitations of target classification from audible sonar images.

11:40

1aAAa4. Investigate echolocation with non-disabled individuals. Alessia Tonelli, Luca Brayda, and Monica Gori (Istituto Italiano di Tecnologia, Via Melen, Genoa 16152, Italy, alessia.tonelli@iit.it)

Vision is the most important sense on the domain of spatial perception. Congenital blind individuals, that cannot rely on vision, show impairments in performing complex spatial auditory tasks. The echolocation technique allows blind people to compensate the audio spatial deficit. Here, we present an overview of our works. First, we show that also sighted people can acquire spatial information through echolocation, i.e., localize an aperture or discriminate the depths of an object locate in front of them. Second, we identified some kinematic variables that can predict the echolocation performance. Third, we show that echolocation, not only helps to understand the external space, but can influence internal models of the body-space relation, such as the peripersonal space (PPS). We discuss all these aspects showing that human beings are sensitive to echoes. Spatial information can be acquired by echolocation when vision is not available also in people that normally would acquire the same information through it. We finally discuss our results in term of rehabilitation technique for visually impaired people.

12:00

1aAAa5. Restoring an allocentric reference frame in blind individuals through echolocation. Tiziana Vercillo (Psych., Univ. of Nevada, Reno, 1664 N. Virginia St., Reno, NV 89503, tvercillo@unr.edu), Alessia Tonelli (U-VIP Unit for Visually Impaired People, Fondazione Istituto Italiano di Tecnologia, Genoa, Italy), Melvyn Goodale (The Brain and Mind Inst., London, ON, Canada), and Monica Gori (U-VIP Unit for Visually Impaired People, Fondazione Istituto Italiano di Tecnologia, Genoa, Italy)

Recent psychophysical studies have described task-specific auditory spatial deficits in congenitally blind individuals. We investigated auditory spatial perception in congenitally blind children and adults during different auditory spatial tasks that required the localization of brief auditory stimuli with respect to either external acoustic landmarks (allocentric reference frame) or their own body (egocentric reference frame). Early blind participants successfully represented sound locations with respect to their body. However, they showed relative poor precision when compared to sighted participants during the localization of sound with respect to external auditory landmarks, suggesting that vision is crucial for an allocentric representation of the auditory space. In a separate study, we tested three congenitally blind individuals who used echolocation as a navigational strategy, to assess the benefit of echolocation on auditory spatial perception. Blind echolocators did not show the same impairment in auditory spatial localization reported for blind non-echolocators, but rather proved enhanced precision and accuracy as compared to blind non-echolocators and sighted participants. Our results suggest that echolocation can compensate for the spatial deficit reported in early blind individuals, likely by reactivating an allocentric reference frame needed to shape spatial representations similar to those generated by vision.

Session 1aAAb

Architectural Acoustics: Sound Propagation Modeling and Spatial Audio for Virtual Reality I

Dinesh Manocha, Cochair
Computer Science, University of North Carolina at Chapel Hill, 250 Brooks Building, Columbia Street, Chapel Hill, NC 27599-3175

Lauri Savioja, Cochair
Department of Media Technology, Aalto University, PO Box 15500, Aalto FI-00076, Finland

U. Peter Svensson, Cochair
Department of Electronic Systems, Norwegian University of Science and Technology, Acoustics Research Centre, Trondheim NO - 7491, Norway

Chair's Introduction—10:35

Invited Papers

10:40

1aAAb1. Experience with a virtual reality auralization of Notre-Dame Cathedral. Brian F. Katz (Lutheries - Acoustique - Musique, Inst. ∂'Alembert, UPMC/CNRS, ∂'Alembert, boîte 162, 4, Pl. Jussieu, Paris 75252 Cedex 05, France, brian.katz@upmc.fr), Barteld N. Postma (LIMSI, CNRS, Université Paris-Saclay, Orsay, France), David Poirier-Quinot (Espaces acoustiques et cognitifs, UMR STMS IRCAM-CNRS-UPMC, Paris, France), and Julie Meyer (LIMSI, CNRS, Université Paris-Saclay, Paris, France)

As part of the 850-year anniversary of Notre-Dame cathedral, Paris, there was a special performance of "La Vierge." A close-mic recording of the concert was made by the Conservatoire de Paris. In an attempt to provide a new type of experience, a virtual recreation of the performance using these roughly 45 audio channels was made via auralization. A computational acoustic model was created and calibrated based on in-situ measurements for reverberation and clarity parameters. A perceptual study with omnidirectional source and binaural receiver validated the calibrated simulation for the tested subjective attributes of reverberation, clarity, source distance, tonal balance, coloration, plausibility, ASW, and LEV when compared to measured responses. Instrument directivity was included for each track's representative orchestral section based on published data. Higher-Order Ambisonic (3rd order) RIRs were generated for all source and receiver combinations using the CATT-Acoustic TUCT software. Virtual navigation throughout a visual 3D rendering of the cathedral during the concert was made possible using an immersive rendering architecture with BlenderVR, MaxMSP, and Oculus Rift HMD. We present major elements of this project: calibration, perceptual study, system architecture, lessons learned, and technological limits encountered with regards to such an ambitious undertaking. [Previously presented in part at EuroRegio2016 & FISM2016.]

11:00

1aAAb2. Bidirectional sound transport. Chunxiao Cao, Zhong Ren (State Key Lab of CAD&CG, Zhejiang Univ., 423 Mengminwei Bldg., Zijingang Campus, Zhejiang University, 866 Yuhangtang Rd., Hangzhou 310058, China, zhongren@acm.org), Carl Schissler, Dinesh Manocha (Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Kun Zhou (State Key Lab of CAD&CG, Zhejiang Univ., Hangzhou, China)

We present a new sound propagation algorithm, Bidirectional Sound Transport (BST), based on bidirectional path tracing. Current state-of-the-art geometric acoustic method handles diffuse reflection by backward path tracing and uses diffuse rain to improve the validity of generated paths. We show that this can be viewed as a special case of bidirectional path tracing. By allowing the connections to be established between any nodes of the subpaths, we are able to improve the sampling quality when sound sources locate near scene objects. This ensures more stable rendering quality and eases ray budget selection. We propose a new metric based on the signal-to-noise (SNR) of the energy response to evaluate the performance of Monte-Carlo path tracing method for sound. Based on the metric, we develop an iterative algorithm to redistribute the samples among bounce numbers according to the statistic characteristics of the sampling of previous frames. We show that the sample redistribution algorithm converges and better balances between early and late reverberation. We evaluate our approach on different benchmarks and demonstrate significant speedup over prior geometric acoustic algorithms. We also discuss clustering algorithms used to improve the scalability for bidirectional sound transport.

11:20

1aAAb3. Efficient construction of the spatial room impulse response. Carl Schissler (Comput. Sci., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Peter Stirling (Oculus, Seattle, WA), and Ravish Mehra (Oculus, 8747 148th Ave. NE, Redmond, WA 98052, ravish.mehra@oculus.com)

An important component of the modeling of sound propagation for virtual reality (VR) is the spatialization of the room impulse response (RIR) for directional listeners. This involves convolution of the listener's head-related transfer function (HRTF) with the RIR to generate a spatial room impulse response (SRIR) which can be used to auralize the sound entering the listener's ear canals. Previous approaches tend to evaluate the HRTF for each sound propagation path, though this is too slow for interactive VR latency requirements. We present a new technique for computation of the SRIR that performs the convolution with the HRTF in the spherical harmonic (SH) domain for RIR partitions of a fixed length. The main contribution is a novel perceptually driven metric that adaptively determines the lowest SH order required for each partition to result in no perceptible error in the SRIR. By using lower SH order for some partitions, our technique saves a significant amount of computation and is almost an order of magnitude faster than the previous approach. We compared the subjective impact of this new method to the previous one and observe a strong scene-dependent preference for our technique. As a result, our method is the first that can compute high-quality spatial sound for the entire impulse response fast enough to meet the audio latency requirements of interactive virtual reality applications.

11:40

1aAAb4. Triton: Practical pre-computed sound propagation for games and virtual reality. Nikunj Raghuvanshi (Microsoft Res., 1 Microsoft Way, Redmond, WA 98052, nikunjr@gmail.com), John Tennant (The Coalition Studio, Microsoft Canada, Vancouver, BC, Canada), and John Snyder (Microsoft Res., Redmond, WA)

Triton is a pre-computed wave-based acoustics system recently shipped in the game "Gears of War 4." Games and VR present exciting new opportunities for virtual acoustics by providing the player with scene-dependent reverberation cues and conveying information about visually occluded areas. Several technical challenges must be met. Scenes containing millions of polygons are common, with mixed indoor-outdoor spaces like broken buildings, courtyards, caves, and rocks. A viable technique must handle this complex visual geometry without users' intervention. The emphasis is on ensuring the resulting auralization is perceptually convincing, varying smoothly on source and listener motion in such scenes. Highly occluded cases with salient paths undergoing multiple edge-diffraction and scattering are common. Computational requirements are quite stringent. A fraction of a single CPU core must be used for acoustic calculations for many tens of moving sources. We discuss how these challenges shape Triton's design. Pre-computation is used to minimize runtime cost. Wave simulation provides complete automation for complex scene geometry. The produced fields contain billions of responses that take terabytes of memory. A key contribution is compact encoding of this data in less than hundred megabytes: objective room acoustic parameters are approached from a novel perspective to aid in spatial compression. The resulting parametric framework is fast and practical for current games and VR applications. Video demonstrations will be shown.

12:00

1aAAb5. Graphical processing units (GPU)-accelerated acoustic simulation for interactive experiences. Tony Scudiero (NVIDIA, 4363 Hamilton Dr., Eagan, MN 55123, tscudiero@gmail.com)

The importance of incorporating acoustic effects to the quality of immersion in virtual reality experiences has been the subject of considerable attention recently due to a resurgence of interest in virtual reality. This work discusses advantages and challenges of using graphical processing units (GPU) in real-time ray-based acoustic simulations for interactive applications, especially virtual reality. Existing ray-tracing libraries such as NVIDIA's OptiX library can be used to create interactive-time simulations which can be applied to audio for virtual reality experiences and games. This work additionally discusses some of the challenges present in creating an accessible library which aims to allow non-experts to easily make use of acoustic simulations to enhance auditory immersion in new virtual reality experiences and games.

Session 1aAAc

Architectural Acoustics: Teaching and Learning in Healthy and Comfortable Classrooms I

Arianna Astolfi, Cochair
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Viveka Lyberg-Åhlander, Cochair
Clinical Sciences, Lund, Logopedics, Phoniatrics and Audiology, Lund University, Scania University Hospital, Lund S-221 85, Sweden

David S. Woolworth, Cochair
Oxford Acoustics, 356 CR 102, Oxford, MS 38655

Invited Papers

10:40

1aAAc1. Active learning in modern schools. Markku Lang (Faculty of Education, Univ. of Oulu, Kaitoväylä 7, Oulu, Oulu 90570, Finland, markku.lang@oulu.fi)

We know that world is changing and in near future workforce is going to be more independent, contingent and temporary. Therefore future learning environments and learning activities are essential to prepare all students for the challenges of work and life. Finland started this school year with a new core curriculum for basic education, which is focusing on developing future skills (developing the key competences as it is described in curriculum). To train and to develop future skills in schools new methods, learning landscapes and activities are needed. Future Classroom Network (European Schoolnet) is using activities as a guideline for creating Learning Zones. These activities and Learning zones are supporting key future skills: Critical Thinking—Investigate Creativity—Create Collaboration—Exchange Learning to learn—Develop Digital competences—Interact Communication—Present But how do these future learning activities look and sound like? What are teachers and students doing, when they are training Future skills? How school design should respect it?

11:00

1aAAc2. The effect of different acoustical treatment in a classroom. Erling Nilsson (Saint-Gobain Ecophon, Box 500, Hyllinge SE-26061, Sweden, erling.nilsson@ecophon.se)

A common room acoustic measure in classrooms and other common public spaces is a suspended sound absorbing ceiling. However, the acoustical condition in the classroom not only depends on the suspended ceiling. The size and shape of the room as well as the properties of building material and the interior fittings and furniture will also affect the room acoustical conditions. Another circumstance is the non-fulfillment of the conditions for the classic diffuse field theory in rooms with absorbent ceiling treatment due to the non-uniform distribution of the absorbing material. This makes the calculation of room acoustic parameters more complex. This paper addresses the effect of different factors that are of importance for the acoustical conditions in a classroom. Outgoing from the unfurnished classroom without suspended ceiling the effect of introducing a suspended ceiling, adding furniture, adding wall panels, as well as extra low frequency absorption will be exemplified based on measurements in a full scale classroom. The room acoustic parameters that are analyzed are the reverberation time T_{20}, the Speech Clarity C_{50} and the Sound Strength G. A calculation model adapted for the non-diffuse conditions in rooms with ceiling treatment will be briefly mentioned.

11:20

1aAAc3. Optimal classroom acoustic design with sound absorption and diffusion for the enhancement of speech intelligibility. Giuseppina E. Puglisi, Filippo Bolognesi, Louena Shtrepi (Dept. of Energy, Politecnico di Torino, Torino, Italy), Anna Warzybok, Birger Kollmeier (Medizinische Physik and Cluster of Excellence Hearing4All, Carl von Ossietzky Universität Oldenburg, Oldenburg, Germany), and Arianna Astolfi (Dept. of Energy, Politecnico di Torino, Corso DC degli Abruzzi, 24, Turin 10124, Italy, arianna.astolfi@polito.it)

Classroom design should be focused on the enhancement of the acoustic comfort for students and teachers. Long reverberation times and excessive noise levels can raise vocal effort and negatively affect speech intelligibility. Recent studies and standards updates have investigated whether acoustic treatment should include both absorbent and diffusive surfaces to account for the teaching and learning premises at the same time; however, studies under realistic conditions for the improvement of existing classrooms acoustics are still needed. In this work, an existing Italian classroom with poor acoustics was considered. Several solutions for treatment were simulated using CATT-Acoustics®, including adjustment of the absorption and scattering coefficients of surfaces differently configured to reach optimal reverberation time, and to increase Speech Transmission Index and Definition, especially for the positions in the furthest raw. The effectiveness of the acoustic treatment was also evaluated in terms of enhancement of speech intelligibility using the Binaural

Speech Intelligibility Model (Rennies *et al*., 2013). Its outcomes are given as speech reception thresholds to yield a fixed level of speech intelligibility. Model predictions indicted an improvement in speech reception thresholds up to 6.8 dB after the acoustic intervention.

11:40

1aAAc4. Good acoustics for teaching and learning. Jonas Christensson (Saint-Gobain Ecophon, St. Gobain Ecophon AB, Box 500, Hyllinge 26503, Sweden, jonas.christensson@ecophon.se)

It is important that classrooms provide good speech intelligibility and speak comfort. Being able to listen without effort is important for learning and we know that poor room acoustics is a burden that impedes learning and affect teachers' voice health. A good classroom is the Swedish forests where we can communicate over long distances without having to raise our voice. I have made several listening tests in forests and also measured the sound reflections in different forests. The results are interesting and I mean that "forest acoustics" should be the goal in terms of acoustic conditions in our schools. Many national sound standards put requirements on room acoustics in classrooms. One requirement is reverberation time, according to ISO 3382-2, and it is often evaluated with $T_{20.}$ Unfortunately, this is a very blunt measure, because we start T_{20}-evaluation first after the sound pressure level dropped 5 dB. This "waiting time" is often quite long and it is a problem because we miss a lot of important information from the early part of the decay curve. Therefore, I mean we have to add C_{50} according to ISO 3382-1, to control if the room acoustics is good enough for teaching.

12:00

1aAAc5. Classroom acoustics and children's speech perception. Lori Leibold (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68124, lori.leibold@boystown.org), Ryan W. McCreery (Audiol., Boys Town National Res. Hospital, Omaha, NE), and Emily Buss (Otolaryngology/Head and Neck Surgery, Univ. of North Carolina, Chapel HIll, NC)

Children must learn in classrooms that contain multiple sources of competing sounds. While there are national standards aimed at creating classroom environments that optimize speech intelligibility (e.g., ANSI/ASA 2010), these standards are voluntary and many unoccupied classrooms fail to meet the acceptable levels specified. Moreover, little attention has been given to measuring and understanding effects of competing speech on children's performance in the classroom. Data will be presented that describe typical noise levels in the classroom. Results from experiments investigating the consequences of competing noise and speech on speech perception at different time points during childhood will be presented. Findings from experiments investigating potential benefits associated with manipulating acoustic cues thought to aid in separating target from background speech will also be discussed.

SUNDAY MORNING, 25 JUNE 2017 ROOM 310, 10:35 A.M. TO 11:40 A.M.

Session 1aAO

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

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

Chair's Introduction—10:35

Invited Paper

10:40

1aAO1. Exploring ocean ecosystems and dynamics through sound. Jennifer L. Miksis-Olds (School of Marine Sci. & Ocean Eng., Univ. of New Hampshire, 24 Colovos Rd., Durham, NC 03824, j.miksisolds@unh.edu)

Acoustic signals propagate long distances in the ocean and provide a means for marine life and humans to gain information about the environment and for marine animals to exchange critical information. Innovation in underwater acoustic technology now permits the remote monitoring of marine life and the environment without the need to rely on human observers, the physical presence of an observation vessel, or adequate visibility and sampling conditions. Passive recordings of the underwater soundscape provide information to better understand the influence of environmental parameters on local acoustic processes, to assess habitat quality and health, and to better understand the risks of ocean noise on marine life. Active acoustic technology provides a high-resolution measure of biological and physical oceanographic processes through time series of backscatter measurements. The ability to obtain passive and active acoustic measurements contemporaneously, along with ancillary data to validate and enhance interpretations, is a powerful tool facilitating insight into ocean and ecosystem dynamics. Knowledge gained and questions raised from the integration of acoustic and oceanographic data in rapidly changing environments will be shared, along with a preview of the Atlantic Deepwater Ecosystem Observatory Network (ADEON) program being launched off the South Atlantic Outer Continental Shelf.

Session 1aBAa

Biomedical Acoustics: Beamforming and Image Guided Therapy I: Algorithms

Costas Arvanitis, Cochair
Mechanical Engineering and Biomedical Engineering, Georgia Institute of Technology, 901 Atlantic Dr. NW, Room 4100Q, Atlanta, GA 30318

Constantin Coussios, Cochair
Institute of Biomedical Engineering, Department of Engineering Science, University of Oxford, Old Road Campus Research Building, Oxford OX3 7DQ, United Kingdom

Invited Paper

10:40

1aBAa1. Frequency-domain passive cavitation imaging. Kevin J. Haworth (Univ. of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209, kevin.haworth@uc.edu), Kenneth B. Bader (Radiology, Univ. of Chicago, Chicago, IL), Kyle T. Rich, Christy K. Holland, and T. Douglas Mast (Univ. of Cincinnati, Cincinnati, OH)

Apfel's three golden rules (know thy sound field, know thy liquid, and know when something happens) should be considered when monitoring acoustic cavitation-based ultrasound therapies. The third rule is often followed using passive cavitation detection with a single-element transducer. However, therapy guidance demands monitoring cavitation activity in the entire tissue volume of interest. Using array-based passive cavitation detection with appropriate beamforming, maps of cavitation activity can be superimposed on pulse-echo, grayscale images of tissue anatomy. In this talk, we will discuss one approach for generating cavitation activity maps, frequency-domain passive cavitation imaging (FD-PCI). FD-PCI implements a delay, sum, and integrate algorithm, which will be described conceptually and mathematically. The advantages and limitations of the algorithm will be discussed in the context of examples. Advantages of FD-PCI include the innate frequency selectivity of the algorithm, the ability to use parallel computing for increased processing speed, the independence of the image resolution from the therapy insonation pulse shape, and the ability to quantify the acoustic power of emissions detected by the array. Challenges of the algorithm will also be discussed, including poor axial resolution and limitations of estimating the emitted acoustic power.

Contributed Papers

11:00

1aBAa2. Optimal beamforming using higher order statistics for passive acoustic mapping. Erasmia Lyka, Christian Coviello, and Constantin Coussios (Dept. of Eng. Sci., Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. Campus Res. Bldg., Headington, Oxford OX3 7DQ, United Kingdom, erasmia.lyka@eng.ox.ac.uk)

Passive Acoustic Mapping (PAM) of sources of nonlinear acoustic emissions has been extensively investigated for monitoring ultrasound therapies. Optimal data-adaptive beamforming algorithms, such as Robust Capon Beamformer (RCB), were readily proposed as a means of improving source localization, accounting simultaneously for array configuration and calibration errors. RCB, however, assumes that signal samples follow a Gaussian distribution. Aiming at improving the spatial resolution of PAM, especially in the axial direction with respect to the array, we propose an alternative beamforming approach, Robust Beamforming by Linear Programming (RLPB). This method makes no assumptions on the statistical distribution of the received signals, and exploits not only the variance but also higher-order-statistics (HOS) of the received signals. Performance evaluation on simulated and *in vitro* experimental data suggests improvement in spatial resolution on the order of 20% and 15% in the axial and transverse directions respectively. This facilitates real-time mapping of disjoint cavitating regions over biologically relevant lengthscales on the order of 2 mm in the axial direction. It is expected that the proposed beamforming approach will provide the necessary improvement in PAM spatial resolution required in several clinically relevant situations, where a single array is used and the ratio of depth to aperture becomes large.

11:20

1aBAa3. Attenuation estimation using passive acoustic mapping. Michael Gray and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Oxford OX37DQ, United Kingdom, michael.gray@eng.ox.ac.uk)

Passive acoustic mapping (PAM) techniques have been developed in order to reduce risk and improve treatment efficacy by localizing and quantifying cavitation emissions during therapeutic ultrasound procedures. The performance of these techniques may be significantly degraded by attenuation between the internal therapeutic target and the external monitoring system. Attenuation itself is an essential parameter in the determination of therapeutic outcomes and safety of treatments such as HIFU ablation or volumetric hyperthermia. However, the spatial and temporal distributions of this parameter are not typically known in clinical scenarios. To address these challenges, we present a method for estimating attenuation using broadband cavitation emissions, potentially allowing for restoration of PAM performance, improved treatment monitoring and guidance, and mapping of tissue attenuation over the course of a treatment. Results from simulations and flow phantom experiments illustrate: (1) the impact of soft tissue-like attenuation on PAM images, (2) the ability to estimate attenuation from

cavitation data, and (3) the enhancement of cavitation source imaging and energy estimation following PAM input data attenuation compensation. In the future, the technique could be expanded as a general broadband method of attenuation correction for conventional diagnostic ultrasound images and improved therapeutic ultrasound treatment planning.

11:40

1aBAa4. Passive acoustic mapping in aberrating media with the angular spectrum approach. Scott J. Schoen (Mech. Eng., Georgia Inst. of Technol., 10,000 Burnet Rd., Austin, TX 78758, scottschoenjr@gatech.edu) and Costas Arvanitis (Mech. Eng., Georgia Inst. of Technol., Boston, MA)

The ability to localize and characterize ultrasound-induced microbubble oscillations through the intact skull with high spatial and temporal resolution holds significant promise for the diagnosis and treatment of brain diseases and disorders. In this study, we investigated the ability of angular spectrum (AS) method, a fast planar projection method, to perform passive acoustic mapping of microbubbles through an intact skull. Finite-difference time-domain numerical simulations were used to model microbubble emissions' propagation through homogeneous, stratified, and 2D inhomogeneous (skull) environments approximately 80 mm by 160 mm. Reconstructions with the AS approach were performed with constant and effective sound speeds, as well as with multi-step propagation, to evaluate their ability to correct for induced aberrations and localize the microbubbles. We also investigated the impact of the receiver position on the localization accuracy. Results for skull simulations indicated that the multi-step AS method reduced the error in axial localization of the microbubbles by on the order of 50% compared with the effective sound speed method, while incurring approximately a 25% increase in computation time for each doubling of the number of propagation steps. Both AS methods were several orders of magnitude faster than time-domain reconstruction. Further investigation of the potential of this approach to correct skull aberrations is warranted.

SUNDAY MORNING, 25 JUNE 2017 — ROOM 312, 10:40 A.M. TO 12:20 P.M.

Session 1aBAb

Biomedical Acoustics: Imaging I

Parag V. Chitnis, Chair

Department of Bioengineering, George Mason University, 4400 University Drive, 1G5, Fairfax, VA 22032

Contributed Papers

10:40

1aBAb1. Ultrasound enhanced delivery of cisplatin loaded nanoparticles. Richard J. Browning, Shuning Bian (Dept. of Eng. Sci., Univ. of Oxford, BUBBL, IBME, ORCRB, Oxford OX3 7DQ, United Kingdom, richard.browning@eng.ox.ac.uk), Philip J. Reardon (Div. of BioMater. and Tissue Eng., UCL Eastman Dental Inst., Univ. College London, London, United Kingdom), Maryam Parhizkar (Mech. Eng., Univ. College London, London, United Kingdom), Anthony H. Harker (Dept. of Phys. & Astronomy, Univ. College London, London, United Kingdom), Vessela Vassileva (Dept. of Oncology, Univ. College London, London, United Kingdom), Dan Daly (Lein Appl. Diagnostics, Reading, United Kingdom), Barbara R. Pedley (Dept. of Oncology, Univ. College London, London, United Kingdom), Mohan Edirisinghe (Mech. Eng., Univ. College London, London, United Kingdom), Jonathan C. Knowles (Div. of BioMater. and Tissue Eng., UCL Eastman Dental Inst., Univ. College London, London, United Kingdom), and Eleanor P. Stride (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Cisplatin forms the basis for many chemotherapy regimens, however the maximum permissible dose is limited by its systemic toxicity. Nanoencapsulation of drugs has been shown to reduce off-target side effects and can potentially improve treatment burden on patients. However, uptake of nanoformulations at tumor sites is minimal without some form of active delivery. We have developed a submicron, polymeric nanoparticle based on biocompatible and degradable poly(lactic-co-glycolic acid) (PLGA) capable of encapsulating cisplatin and which can be bound to the surface of a phospholipid coated microbubble. The acoustic behavior and stability of the resulting nanoparticle loaded microbubbles will be compared with those of unloaded microbubbles. Results will also be presented on the extravasation of particles in a tissue mimicking phantom using a novel long working distance confocal microscope that enables particle distributions to be measured in situ and in real time.

11:00

1aBAb2. Optimizing gold nanorod volume for minimum cell toxicity and maximum photoacoustic response. Oscar B. Knights, David Cowell (School of Electron. & Elec. Eng., Univ. of Leeds, Leeds LS2 9JT, United Kingdom, elok@leeds.ac.uk), James R. McLaughlan (Div. of Biomedical Imaging, Univ. of Leeds, Leeds, United Kingdom), and Steven Freear (School of Electron. & Elec. Eng., Univ. of Leeds, Leeds, United Kingdom)

Plasmonic nanoparticles show great potential for molecular-targeted photoacoustic (PA) imaging. To maximize light absorption, the gold nanorods (AuNRs) are illuminated at their surface plasmon resonance (SPR), which for biomedical application is typically in the "optical window" of 700-900 nm. For AuNRs, one of the main factors that determines the SPR is their aspect ratio. Since it is possible to have a similar aspect ratio, but different size of the particle the choice of particle could have a critical effect on a number of factors, such as photoacoustic emissions, cell toxicity, and therapeutic efficacy. For example, a particular sized AuNR may produce a higher PA response, for an equivalent laser fluence, but be more toxic to cell populations. In this study, the PA response of AuNRs with four different volumes but similar aspect ratios (~4) are compared. A linear relationship between incident laser fluence and PA amplitude is shown and results indicate that AuNRs with larger volumes produce stronger PA emissions. *In-vitro* cell studies were performed on a lung cancer cell line to assess the cell toxicity of the different sized AuNRs via a colorimetry assay.

11:20

1aBAb3. Ultrasound-mediated blood-brain barrier disruption: Correlation with acoustic emissions. Miles M. Aron, Lester Barnsley, Shamit Shrivastava (Dept. of Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg., IBME, Roosevelt Dr., Oxford, Oxfordshire OX3 7DQ, United Kingdom, Miles.aron@hertford.ox.ac.uk), Marinke Van der Helm, Loes Segerink (Faculty of Elec. Eng., Mathematics and Comput. Sci., Univ. of Twente, Enschede, Netherlands), and Eleanor P. Stride (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Blood-brain barrier (BBB) disruption mediated by ultrasound and microbubbles (US-BBBD) is a promising strategy for non-invasive and targeted delivery of therapeutics to the brain. In US-BBBD, treatment control is achieved by externally monitoring acoustic emissions (AE) and adjusting ultrasound parameters in real-time to avoid AE associated with damage. Recent work suggests that AE may also provide insight regarding the extent of BBB opening and BBB recovery time. The mechanisms underlying BBB opening and recovery, however, are largely not understood. To investigate US-BBBD mechanisms with regard to AE, we developed an *in vitro* platform for monitoring both BBB integrity and AE during US-BBBD. Temporally resolved BBB integrity monitoring was achieved using a microfluidic BBB-on-a-chip device with integrated trans-endothelial electrical resistance (TEER) measurements. Well-characterized ultrasound exposure and AE monitoring were achieved using a focally aligned high-intensity focused ultrasound transducer and passive cavitation detector. In addition to recording TEER and AE data, our platform is compatible with fluorescence microscopy during ultrasound exposure, providing further insight into US-BBBD mechanisms. This work further demonstrates potential for *in vitro* screening of cavitation agents and/or therapeutics for novel US-BBBD applications and strategies.

11:40

1aBAb4. Nonlinear propagation of two dimensional sound waves observed at lipid interfaces. Shamit Shrivastava (Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, shamit.shrivastava@eng.ox.ac.uk) and Matthias F. Schneider (Medizinische und biologische Physik, Technische Universitat, Dortmund, Germany)

Experimental results are presented on the acoustic propagation of mechanical perturbations in a lipid monolayer along the air-water interface. The interface was excited by a piezo-cantilever, and propagating impulses were measured optically using Forster Resonance Energy Transfer (FRET). The velocity of propagation varied from 0.1 to 1 m/s depending on the compressibility of the interface. Near a nonlinearity in the state diagram of the interface, for example, near a phase transition of the lipids, impulses propagated only when the initial mechanical impulse was greater than a certain threshold. In fact, the impulse then propagated as a solitary shock wave causing local adiabatic phase transition of the lipid molecules along the way. Although the phenomenon has been observed here in pure lipid monolayers, which represent a well-documented model for biological membranes, the origin of the observed phenomenon lies in the conservation of the entropy of the interface, determined by the change in state of the interface during the impulse. Given that the state diagrams of biological membranes have nonlinearities near physiological conditions, nonlinear sound waves are expected to be fundamentally involved in inter and intra cellular communication. Indeed, the observed acoustic phenomenon is characteristically similar to nerve impulses.

12:00

1aBAb5. Focus ultrasound for augmenting convection-enhanced delivery of nanoparticles in the brain. Ali Mohammadabadi (Diagnostic Radiology and Nuclear Medicine, Univ. of Maryland School of Medicine, 110 S. Paca St., Rm. 104, Baltimore, MD 21201, ali.mohammadabadi@umm.edu), David S. Hersh (Neurosurgery, Univ. of Maryland School of Medicine, Baltimore, MD), Pavlos Anastasiadis (Diagnostic Radiology and Nuclear Medicine, Univ. of Maryland School of Medicine, Baltimore, MD), Philip Smith, Graeme F. Woodworth, Anthony J. Kim (Neurosurgery, Univ. of Maryland School of Medicine, Baltimore, MD), and Victor Frenkel (Diagnostic Radiology and Nuclear Medicine, Univ. of Maryland School of Medicine, Baltimore, MD)

We previously demonstrated how ultrasound can enhance the dispersion of locally administrated nanoparticles within the extracellular/perivascular spaces in the *ex vivo* brain by non-destructively enlarging these regions. The current study aimed to translate these results *in vivo*, where custom, non-adhering brain-penetrating nanoparticles (BPN: 60, 200, and 500 nm), were administered directly into the brains of Sprague Dawley rats by convection-enhanced delivery. Non-invasive, transcranial focused ultrasound (TCFUS) was carried out using an MRI-guided system (1.5 MHz, 10 ms pulses, 10% duty cycle, and 2.3 MPa). 15 individual exposures in a 3×5 matrix (spacing: 1.5 mm) in one hemisphere were given, where the size of the focal zone (-6 dB) was 1×1 X 8 mm. At 2hrs post-treatment brains were harvested and sectioned, with digital images captured and processed using a custom MATLAB script. This involved the "Otsu" thresholding method, based on gray level histograms and threshold determinations for maximizing the interclass variance. As expected, BPN distributions in the non-treated brains decreased with an increase in diameter. Pretreating with TCFUS was found to significantly increase the distribution of the 200 nm BPNs. These results have broad implications for therapeutic delivery for a variety of brain diseases and disorders.

Session 1aNS

Noise, Physical Acoustics, ASA Committee on Standards, and Structural Acoustics and Vibration: Sonic Boom Noise I: Low Boom Technology, Propagation, Etc.

Philippe Blanc-Benon, Cochair
Centre acoustique, LMFA UMR CNRS 5509, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, Ecully 69134 Ecully Cedex, France

Victor Sparrow, Cochair
Grad. Program in Acoustics, Penn State, 201 Applied Science Bldg., University Park, PA 16802

Chair's Introduction—10:35

Invited Papers

10:40

1aNS1. Status and plans for NASA's Quiet SuperSonic Technology (QueSST) aircraft design. Peter Coen and David Richwine (NASA, NASA Langley Res. Ctr., MS 264, Hampton, VA 23681, peter.g.coen@nasa.gov)

Innovation in Commercial Supersonic Technology is one of six thrusts that guide NASA's Aeronautics Research Strategy. The near term objective of this activity is the establishment of a standard for acceptable overland supersonic flight, in cooperation with international standards organizations. In support of this objective, NASA supersonics research has had two focus areas in recent years. The first is the design of aircraft that can fly at supersonic speeds without creating a loud sonic boom, and the second is to understand the community response to the relatively quiet sound of the overflight of such aircraft. Based on the recent successes in this research, NASA has determined that the next steps in both these areas and in continued progress toward the near term objective is a flight demonstration. NASA, in cooperation with industry partners, has initiated the preliminary design of a Low Boom Flight Demonstration Aircraft, named QueSST for Quiet Supersonic Technology. This paper will describe the development of the design requirements for QueSST, and provide an overview of the design progress to date and future plans for the Flight Demonstration Project.

11:00

1aNS2. Development of high fidelity tools and robust design approaches for low boom aircraft. Lori Ozoroski (ASAB, NASA, 1 North Dryden St., Hampton, VA 23681, lori.p.ozoroski@nasa.gov) and Linda Bangert (CAB, NASA, Hampton, VA)

The NASA Commercial Supersonic Technology Project recently completed a project Technical Challenge, "Low Sonic Boom Design Tools" which ran from 2011 to 2015. As part of this research effort, tools were developed, refined, and validated to support full-vehicle low-boom analysis, including inlet and nozzle effects. In addition, new and updated tools and processes were developed and demonstrated for application to robust low boom shape optimization. The work included both fundamental research efforts, computational analysis of full vehicle configurations, and application of robust low boom design methods to low boom aircraft concepts. This presentation will primarily focus on the technical achievements during the last 2 years leading to the recent successful completion of this technical challenge.

11:20

1aNS3. Advances in numerical simulation of sonic boom in realistic atmospheres. François Coulouvrat, David Luquet, Régis Marchiano (Institut Jean Le Rond d'Alembert (UMR 7190), Universite Pierre et Marie Curie & CNRS, Université Pierre et Marie Curie, 4 Pl. Jussieu, Paris 75005, France, francois.coulouvrat@upmc.fr), and Franck Dagrau (Dassault Aviation, Saint Cloud, France)

Efficient and accurate numerical simulation of sonic boom is a key issue for both the design of low boom aircraft and the definition of a standard on supersonic overland flights. Atmospheric turbulence is known since the 1960s to significantly alter the ideal N-wave boom waveform. Such random fluctuations cannot be reproduced by the standard ray tracing method. Neither can it simulate the lateral boom beyond the geometrical carpet edge. There occur many non-geometrical features such as creeping waves, wave guiding or scattering. To progress in the direction of boom simulation beyond the geometrical approximation, we developed the so-called FLHOWARD3D software. For accuracy, it can handle 3D temperature, density or wind heterogeneities, atmospheric absorption, and nonlinear propagation effects. For efficiency, a one-way approximation is performed, neglecting the backscattered field. Nevertheless, the forward field satisfies an accurate dispersion relation, even in moving atmospheres. Involved physical effects are handled separately by optimized algorithms, combined by a second-order split-step approach. The resulting software is parallelized using the MPI paradigm. The performances of the software will be illustrated by two cases: boom scattering by turbulence, and lateral boom propagation in case of temperature inversion. This last case will be compared with flight test data.

11:40

1aNS4. Progress made during the first two American Institute for Aeronautics and Astronautics Sonic Boom Prediction Workshops for calculating near field signatures of supersonic aircraft. Michael A. Park (Langley Res. Ctr., NASA, NASA Langley Res. Ctr., MS 128, Hampton, VA 23681, Mike.Park@NASA.gov)

The American Institute for Aeronautics and Astronautics (AIAA) Sonic Boom Workshops examine the international state of the art in sonic boom prediction of supersonic aircraft. A summary is provided for the first and second workshops held in 2014 and 2017. The nearfield CFD (Computational Fluid Dynamics) cases from both workshops are described. They include a range of simple to complex configurations. The first workshop used models with N-wave, flat-top, and shaped ground signatures. The second workshop used models with quieter shaped ground signatures. To assess state of the art in nearfield CFD prediction, signatures are gathered from the international participants. These nearfield signatures are propagated to the ground with an augmented Burgers equation method and noise metrics are calculated. Statistics of the noise metrics are utilized to identify outliers for further examination. Comparisons are also made to wind tunnel measurements through validation metrics where available. The convergence of these metrics with grid refinement is also documented. This allows the determination of state of the art for nearfield sonic boom prediction and documents progress made between the two workshops.

12:00

1aNS5. Summary and progress made in modeling of sonic boom propagation during AIAA sonic boom prediction workshops. Sriram Rallabhandi (Aeronautics Systems Anal. Branch, NASA Langley, Rm. 190-25, Mailstop 442, NASA Langley Res. Ctr., Hampton, VA 23681, sriram.rallabhandi@nasa.gov) and Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

This paper summarizes the atmospheric propagation modeling portion of the Second American Institute of Aeronautics and Astronautics (AIAA) Sonic Boom Prediction Workshop held in 2017 as well as an informal propagation comparison and benchmarking effort conducted prior to the First AIAA Sonic Boom Prediction Workshop in 2014. The motivation behind these workshops is the industry's increased interest in low boom supersonic aircraft designs and the need to have an open, unbiased forum to promote best practices. The propagation test cases from both exercises are described and discussed. Discussion is also included on the selection of test case conditions with multiple atmospheric profiles, representing geographical and seasonal variations of the relevant meteorological data. Propagated sonic boom ground signatures, loudness metrics, extent of the boom carpets and other propagation-related details were gathered from a group of international participants. Comparisons are made between submissions, and the differences are analyzed in detail to understand the state-of-the-art in sonic boom atmospheric propagation modeling. The progress made between the workshops and the lessons learned will be discussed.

SUNDAY MORNING, 25 JUNE 2017 ROOM 210, 10:40 A.M. TO 12:20 P.M.

Session 1aPA

Physical Acoustics and Biomedical Acoustics: Acoustofluidics I

Jürg Dual, Cochair
ETH Zurich, Tannenstr. 3, Zurich 8092, Switzerland

Charles Thompson, Cochair
ECE, UMASS, 1 Univ Ave, Lowell, MA 01854

Max Denis, Cochair
U.S. Army Research Lab., 2800 Powder Mill Road, Adelphi, MD 20783-1197

Invited Papers

10:40

1aPA1. Exploring the phenomenon of ultrasonic atomization for viscous fluids. James Friend (Mech. and Aerosp. Eng., Univ. of California, San Diego, 345F Structural and Mech. Eng., M.S. 411 Gilman Dr., La Jolla, CA 92093, jfriend@eng.ucsd.edu)

We consider the choice of vibration modes and piezoelectric materials for acoustically driven atomization, an attractive method for a broad range of applications, particularly pulmonary drug delivery. Whether by the definition of a figure of merit, a product of the resonator quality factor and electromechanical coupling coefficient, its output vibration displacement at a given input power, or the fluid flow

rate during atomization, we find that the combination of single-crystal 127.86-deg. Y-rotated lithium niobate and thickness-mode vibration produces an order of magnitude greater atomization flow rate and efficiency in comparison to classic lead zirconate-based devices and newer, Rayleigh wave or Rayleigh/Lamb spurious-mode based devices alike. For the first time, fluids with viscosities up to 48 cP are reported to be atomized, and we define an atomization Reynolds number Re_A that can be used to both predict the atomization flow rate for $Re_{A>40}$ and the inability to atomize a given fluid at a particular vibration amplitude when $Re_{A<40}$.

11:00

1aPA2. Predicting droplet sizes and production rates in ultrasonic atomization as a strongly nonlinear phenomena. James Friend (Mech. and Aerosp. Eng., Univ. of California, San Diego, 345F Structural and Mech. Eng., M.S. 411 Gilman Dr., La Jolla, CA 92093, jfriend@eng.ucsd.edu)

Atomization of fluids using ultrasonic irradiation of a fluid interface is well-known, yet the underlying physics is surprisingly complex and rather poorly understood. We review classical and modern theories of capillary wave formation and droplet atomization, from Michael Faraday in 1831 onwards, and show how the atomization process occurs via induction of bulk turbulence that in turn engenders capillary wave production and turbulence. We show the clear separation of droplet generation rate from the frequency of the excitation ultrasound, and instead show the existence of a specific atomization frequency dependent upon the ratio of the fluid's surface tension to its viscosity.

11:20

1aPA3. Acoustic radiation force expansions in terms of partial-wave scattering phase shifts: Extended applications. Philip L. Marston (Phys. & Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu) and Likun Zhang (Phys. Dept., Univ. of MS, Oxford, MS)

When evaluating radiation forces on spheres in sound fields, the interpretation of analytical results is greatly simplified by retaining the use of s-function notation for partial-wave coefficients imported into acoustics from quantum scattering theory. This facilitates easy interpretation of various scattering efficiency factors [L. Zhang and P. Marston, J. Acoust. Soc. Am. 140, EL178 (2016)]. This also facilitates the correction of certain plane-wave results [H. Olsen *et al.*, J. Acoust. Soc. Am. 30, 633 (1958)] and the force parameterization for a broader class of wavefields. For situations in which dissipation is negligible, each partial-wave s-function becomes characterized by a single parameter: a phase shift. These partial-wave phase shifts are associated with scattering by plane traveling waves; the incident wavefields of interest (progressive and standing wavefields and beams) are separately parameterized. (When considering outcomes, the method of fabricating symmetric objects having a desirable set of phase shifts becomes a separate issue.) The existence of negative radiation force "islands" for beams reported in 2006 by Marston is manifested. Elementary standing and traveling wave force expressions are also recovered. This approach also manifests the utility of conservation theorems [P. Marston and L. Zhang, J. Acoust. Soc. Am. 139, 3139 (2016)]. [Work supported by ONR.]

11:40

1aPA4. Beyond acoustophoresis: Particle manipulation near oscillating interfaces. Sascha Hilgenfeldt (Mech. Sci. and Eng., Univ. of Illinois, 1206 W Green St., Urbana, IL 61801, sascha@illinois.edu), Bhargav Rallabandi (Mech. and Aerosp. Eng., Princeton Univ., Princeton, NJ), Siddhansh Agarwal, and David Raju (Mech. Sci. and Eng., Univ. of Illinois, Urbana, IL)

Inertial effects in microfluidics afford an interesting set of tools for the control of particle positions. The gradients of steady channel flows, as well as the gradients of acoustic field amplitudes, have been used prominently to this purpose, the latter in acoustofluidics. Here, we investigate directly the effect of an oscillating interface on the fluid surrounding it and particles suspended in the fluid. The fast oscillatory motion gives rise to strong inertial effects, while the method allows for versatile force actuation because of the variety of flow fields, frequencies, and length scales under the experimentalist's control. We show in experiment and theory that oscillating bubbles simultaneously (i) guide particles close to the bubble interface by streaming flow, and (ii) exert strong lift forces that can be used to sort the particles by size or density. The lift forces and the ensuing particle displacement constitute an effect separate from streaming and can be understood analytically on the time scale of oscillation and that of averaged, steady motion; unlike classical acoustofluidics, it does not rely on density or compressibility contrasts. Comparison with experiments confirms that particle displacements scale more favorably and flexibly with the dimensions of particle and microfluidic set-up than in traditional inertial microfluidics. Size sorting with micrometer resolution can be accomplished within a millisecond, while the same device can exert controlled repulsive and attractive forces.

12:00

1aPA5. Beyond acoustophoresis: Attractive and repulsive forces on particles. Sascha Hilgenfeldt (Mech. Sci. and Eng., Univ. of Illinois, 1206 W Green St., Urbana, IL 61801, sascha@illinois.edu), Bhargav Rallabandi (Mech. and Aerosp. Eng., Princeton Univ., Princeton, NJ), Siddhansh Agarwal, and David Raju (Mech. Sci. and Eng., Univ. of Illinois, Urbana, IL)

Inertial effects in microfluidics afford an interesting set of tools for the control of particle positions. The gradients of steady channel flows, as well as the gradients of acoustic field amplitudes, have been used prominently to this purpose, the latter in acoustofluidics. Here, we investigate directly the effect of an oscillating interface on the fluid surrounding it and particles suspended in the fluid. The fast oscillatory motion gives rise to strong inertial effects, while the method allows for versatile force actuation because of the variety of flow fields, frequencies, and length scales under the experimentalist's control. We show in experiment and theory that the forces on particles can be evaluated analytically, on both the oscillatory and the steady, time-averaged time scales. The latter formalism generalizes streaming flow computations to particle motion, and reveals new potential strategies for manipulating particles with tunable attractive or repulsive forces, depending not only on characteristics of the particles and physical properties of the fluid, but also the dynamical parameters of the driving.

Session 1aPPa

Psychological and Physiological Acoustics: Perception of Synthetic Sound Fields I

Sascha Spors, Cochair
Institute of Communications Engineering, University of Rostock, Richard-Wagner-Strasse 31, Rostock 18119, Germany

Nils Peters, Cochair
Advanced Tech R&D, Qualcomm Technologies, Inc., 5775 Morehouse Drive, San Diego, CA 92121

Invited Paper

10:40

1aPPa1. Evaluation of object-based audio—What is the reference? Thomas Sporer, Judith Liebetrau, and Tobias Clauss (Fraunhofer IDMT, Ehrenberg Str. 31, Ilmenau 98693, Germany, spo@idmt.fhg.de)

During the development of audio coding schemes, a number of methods for evaluation of the perceived audio quality have been developed. To enable comparisons across test sites, several methods have been standardized. In standardized methods like ITU-R Recommendations BS.1116 and BS.1534 (MUSHRA), the output of a codec (signal under test) is compared to an open reference. This reference is the unimpaired input of the codec. Assuming that the codec is "transparent," the signal under test should sound exactly like this reference. For object-based audio, the input of a codec is a combination of raw audio channels and metadata describing position and other properties of the audio objects. It does not make sense to listen directly to the raw data. For listening, it is necessary to calculate the driving signal for each loudspeaker available in the listening room (rendering). Therefore, the comparison of different renderers is difficult: the renderer used to generate the reference signal has an advantage. Using a dedicated loudspeaker as the reference does not solve the problem either: loudspeakers always sound different than virtual sound objects. The presentation will discuss problems and solutions in more detail. Some promising setups based on multi attribute testing are presented.

Contributed Papers

11:00

1aPPa2. Investigation of perceptual attributes associated with projected sound sources. Tom Wühle and M. Ercan Altinsoy (Chair of Acoust. and Haptics, Dresden Univ. of Technol., Helmholtzstraße 18, Dresden 01062, Germany, tom.wuehle@tu-dresden.de)

One solution to reproduce sound from various directions is the projection of sound sources on reflective boundaries. In this case, the perceived direction of the auditory event changes from the direction of the real source to the direction of the projected source. Therefore, highly focused sound sources are necessary. However, the focusing capabilities of such sources are physically limited. Thus, the total amount of sound at the listening position is formed by the projected sound and by sound which is directly radiated from the real source. Both of these sound proportions influence the perception of the listener. In a scenario with projected sound sources, a complex mixture of perceptual attributes change besides the direction of the auditory event. The present study investigates some of those attributes.

11:20

1aPPa3. A user-centered taxonomy of factors contributing to the listener experience of reproduced audio. James Woodcock, William J. Davies, and Trevor J. Cox (Acoust. Res. Ctr., Univ. of Salford, Newton Bldg., Salford M5 4WT, United Kingdom, j.s.woodcock@salford.ac.uk)

The traditional paradigm for the assessment of audio quality is that of a listener positioned in the geometric center of a standardized loudspeaker setup, fully attending to the reproduced sound scene. However, this is not how listeners generally interact with audio technology. Audio is consumed in a variety of environments and situations, over devices with varying quality, and by listeners with different expectations and needs. Drawing on research from soundscapes, human computer interaction, and multimedia quality of experience, this paper proposes a user-centered taxonomy of factors that influence the listener experience of reproduced audio. The taxonomy is supported by data from recent research into the perception of complex reproduced sound scenes, and by new data from a web-based survey investigating the structure of experiences with audio technology. In this survey, participants were asked to consider previous experiences with audio technology, and data were collected on the experience itself (psychological need fulfillment and affect), perceptual attributes related to the reproduced audio, and the importance of audio quality to the experience. Results point toward a model of listener experience that can be used to profile listener experiences in different contexts and can be used as a measurement tool in future controlled experiments.

Invited Papers

11:40

1aPPa4. Individual attributes describing the perception of synthesized sound fields in listening rooms. Vera Erbes and Sascha Spors (Inst. of Communications Eng., Univ. of Rostock, Richard-Wagner-Str. 31, Haus 8, Rostock 18119, Germany, vera.erbes@uni-rostock.de)

Sound field synthesis techniques such as Wave Field Synthesis (WFS) are in theory defined in an anechoic environment but real-world installations have to be placed inside listening rooms with reflective walls. The arising reflections influence the desired synthesized sound field both from a physical and a perceptual point of view. This study investigates the perceptual aspects that are relevant for WFS in reflective environments by means of the Repertory Grid Technique (RGT). By comparing a representative range of auditory scenes with real and virtual sources in free field and in different listening rooms, subjects generate individual constructs to describe perceived differences and similarities as contrastive pairs. In a second step, subjects rate the scenes on scales constructed by their own contrastive pairs. The ratings are used to reveal relations between the constructs and the scenes on an individual basis. Clusters of constructs can be found that in terms of content can be associated with a common perceptual aspect.

12:00

1aPPa5. Using binaural and spherical microphone arrays to assess the quality of synthetic spatial sound fields. Jonas Braasch, Nikhil Deshpande, Jonathan Mathews, and Samuel Chabot (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Recently, we completed the Collaborative-Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab) with a usable floor area of 12×10 m^2 at Rensselaer. The CRAIVE-Lab project addresses the need for a specialized virtual-reality (VR) system for the study and enabling of communication-driven tasks with groups of users immersed in a high-fidelity multi-modal environment located in the same physical space. For the acoustic domain, a 134-loudspeaker-channel system has been installed for Wave Field Synthesis (WFS) with the support of Higher-Order-Ambisonic (HoA) sound projection to render inhomogeneous acoustic fields. An integrated 16-channel spherical microphone array makes the CRAIVE-Lab an ideal test bed to study different spatial rendering techniques such as Wave-Field Synthesis, Higher-Order Ambisonics and Virtual Microphone Control (ViMiC). In this talk, sound-field measurements taken with a traditional binaural manikin will be compared to spherical microphone recordings to assess the quality of the different rendering techniques for large-scale labs. A focus will hereby be set on assessing the sweet spot area for different rendering techniques. [Work supported by NSF 1229391, NSF 1631674, and the Cognitive and Immersive Systems Laboratory (CISL).]

SUNDAY MORNING, 25 JUNE 2017 ROOM 311, 10:55 A.M. TO 12:00 NOON

Session 1aPPb

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Andrea Simmons, Chair
Brown University, Box 1821, Providence, RI 02912

Chair's Introduction—10:55

Invited Paper

11:00

1aPPb1. Active listening in 3D auditory scenes. Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., Biology-Psych. Bldg. 2123M, College Park, MD 20742, cynthia.moss@gmail.com)

As an animal moves in its natural environment, to seek food, track targets, and steer around obstacles, its distance and direction to objects continuously change, invoking dynamic feedback between 3D scene representation, attention, and action-selection. Animals that rely on active sensing provide powerful systems to investigate neural underpinnings of sensory-guided behaviors, as they produce the very signals that inform motor actions. Echolocating bats, for example, transmit sonar signals and process auditory information carried by echoes to guide behavioral decisions for spatial orientation. Further, the bat adapts its echolocation signal design in response to 3D

spatial information computed from echo returns, and therefore, the directional aim and temporal patterning of its calls provide a window into the animal's attention to objects in its surroundings. In addition, the bat actively controls pinna position and head movements to enhance auditory cues about 3D target position. These adaptive behaviors require an interface between auditory processing and motor commands, and our research findings implicate the midbrain superior colliculus in sensory-guided spatial orienting behaviors. This talk will review behavioral and neurobiological studies of 3D sonar scene analysis in the echolocating bat, an animal whose active control over acoustic signals provides a window into its perceptual world.

SUNDAY MORNING, 25 JUNE 2017 ROOM 201, 10:35 A.M. TO 12:20 P.M.

Session 1aSA

Structural Acoustics and Vibration, Noise, Physical Acoustics, and ASA Committee on Standards: Groundborne Noise and Vibration from Transit Systems

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

Chair's Introduction—10:35

Invited Papers

10:40

1aSA1. Draft standard on "Methods for the Prediction of Ground Vibration from Rail Transportation Systems." James E. Phillips (Wilson, Ihrig & Assoc., Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

A draft American National Standards Institute (ANSI) standard on "Methods for the Prediction of Ground Vibration from Rail Transportation Systems" is nearing completion for review. The intent of this document is to standardize methods that were initially developed thirty years ago and adopted by the Federal Transit Administration in the guidance manual "Transit Noise and Vibration Assessment" for determining environmental vibration impacts at sensitive land uses adjacent to transit projects. This paper will outline the topics in the draft standard.

11:00

1aSA2. Ground vibration propagation measurements in extreme conditions. Scott Edwards (Senior Associate, Cross-Spectrum Acoust., 1500 District Ave, Ste. 1011, Burlington, MA 01803, sedwards@csacoustics.com)

Ground vibration propagation measurements are often required during the environmental impact assessment process for transit projects per Federal Transit Administration (FTA) guidance. Cross-Spectrum Acoustics (CSA) conducted these measurements in extremely hot (100+ degrees Fahrenheit) and extremely cold (sub-zero degrees Fahrenheit) environments in 2016. This presentation will provide an overview of conducting such procedures in extreme climates and lessons learned from CSA staff. Topics to be discussed will include the effects of extreme temperatures on equipment, staff, and the collected data.

11:20

1aSA3. Ground-borne vibration issues from rail transit near university research buildings. Timothy Johnson and Gary Glickman (Wilson Ihrig, 30 E. 20th St., New York, NY 10003, tjohnson@wiai.com)

A new light rail transit project being designed and constructed through a university campus presents many potential ground-borne vibration issues. The rail alignment passes near numerous buildings on campus that contain various types of vibration sensitive research equipment. Appropriate vibration criteria for the project are critical to ensure that future operation of the light rail vehicles (LRVs) will not adversely affect activities on campus. Ground-borne vibration criteria are based on equipment sensitivity and the existing vibration environment. Projections of future vibration levels from LRV operations were developed based on field measurement programs. Site specific vibration measurements on campus were conducted to document the soil vibration propagation characteristics and building response characteristics. Additionally, input vibration force characteristics, or vehicle force density levels, of a comparable vehicle were measured and incorporated into the projections. Finally, site specific track design and vibration mitigation measures were modeled and incorporated into the project design.

Contributed Papers

11:40

1aSA4. Prediction of ground-borne vibrations on historical structures due to tram traffic in Antalya, Turkey. Salih Alan and Mehmet Caliskan (Dept. of Mech. Eng., Middle East Tech. Univ., Ankara 06800, Turkey, caliskan@metu.edu.tr)

This study investigates the ground-borne vibrations on historically landmarked structures in the city of Antalya due to tram traffic. Assessments are conducted over predicted vibrations with respect to international standards. The study serves as a guidance for the tenders in the upcoming tender process for projects of upgrading existing tram lines as well as construction of new lines. In the prediction procedure, an existing Fourier transform based theoretical model for the track and layered ground is implemented coupling vehicle dynamics. Groundborne vibrations calculated at the base level of structures are considered. Vibration assessment criteria taken from ISO 2631 standard are employed in the evaluations of predicted vibrations at the respective locations in three mutually perpendicular directions.

12:00

1aSA5. Predicting structure-borne sound from railway traffic. Juan Negreira (Eng. Acoust., Lund Univ., Eng. Acoust., LTH, BOX 118, Lund, Skane 22100, Sweden, juan.negreira@construction.lth.se), Peter Persson (Structural Mech., Lund Univ., Lund, Skane, Sweden), and Delphine Bard (Eng. Acoust., Lund Univ., Lund, Sweden)

Since noise exposure can disturb the well-being, acoustical comfort in the built environment is of great importance when constructing new dwellings. Population growth causes densification of cities, which together with space limitation issues, lead to buildings being constructed closer to existing vibration sources such as motorways and railways, and vice versa. Likewise, architectural trends, environmental benefits and cost result in increased use of lighter materials such as wood and hollow-core concrete slabs. Lightweight structures make the achievement of acoustical comfort in dwellings an increasing challenge. A major issue when designing buildings regarded as acoustically pleasant, especially in the low-frequency range, is the lack of reliable prediction models to be used during the design stage of the building. Predictions of structure-borne noise are nowadays mostly made based on measurements performed on existing buildings and engineers' experience. Hence, it is of interest to develop tools that could adequately predict noise and vibrations. The computer models developed for that purpose could combine different numerical methods, and they may use measurement data as input. The aim here is to investigate and develop numerical models that could be used in the early design stage of structures, specially aimed at predicting structure-borne noise in railway tunnels.

SUNDAY MORNING, 25 JUNE 2017 — BALLROOM A, 10:40 A.M. TO 12:20 P.M.

Session 1aSC

Speech Communication: Speech Technology (Poster Session)

Kelly Berkson, Chair
Dept. of Linguistics, Indiana Univ., 1021 E. Third St., Mem 322E, Bloomington, IN 47405

All posters will be on display from 10:40 a.m. to 12:20 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 10:40 a.m. to 11:30 a.m. and authors of even-numbered papers will be at their posters from 11:30 a.m. to 12:20 p.m.

Contributed Papers

1aSC1. How time-based alignment of realized acoustic landmarks and predicted landmarks improves analysis of feature cue modification patterns in speech. Rebekah Bell, Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Res. Lab. of Electronics, Massachusetts Inst. of Technol., 50 Vassar St., Rm. 36-511, Cambridge, MA 02139, bellr@mit.edu)

Acoustic landmarks are abrupt spectral changes that signal the underlying manner features of phonemes (Stevens 2002). Our goal in developing an automatic method to detect these landmarks is to create a robust, knowledge-based approach to phoneme extraction in automatic speech signal processing. One challenge in such an approach is posed by massive reductions, in which many landmarks and other feature cues are missing. Thus, there is a need to hand-label the acoustic landmarks that actually occur in the speech signal and align them with predicted landmarks. However, this often results in a discrepancy between the locations of labels for the automatically generated and hand-labeled landmarks, which leads to an inaccurate analysis of where realized landmarks occur with respect to word and phoneme interval boundaries. We attempted to solve this issue with a time-based alignment method derived from the minimum edit distance algorithm. The result improved alignment of the realized landmark labels with the predicted landmark labels, enabling a more accurate analysis of modifications in the hand-labeled (realized) landmark tiers.

1aSC2. Landmark-based consonant voicing detection on multilingual corpora. Xiang Kong (Comput. Sci., Univ. of Illinois at Urbana Champaign, Champaign, IL), Xuesong Yang, Mark Hasegawa-Johnson (Beckman Inst., Univ. of Illinois at Urbana-Champaign, Urbana, IL), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Speech Commun. Group, Res. Lab. of Electronics, MIT, 50 Vassar St., Rm. 36-581, Cambridge, MA 02139, jyechoi@mit.edu)

This study tests the hypothesis that distinctive feature classifiers anchored at phonetic landmarks can be transferred cross-lingually without loss of accuracy. Three consonant voicing classifiers were developed: (1) manually selected acoustic features anchored at a phonetic landmark, (2) MFCCs (either averaged across the segment or anchored at the landmark), and (3) acoustic features computed using a convolutional neural network (CNN). All detectors are trained on English data (TIMIT) and tested on English, Turkish, and Spanish (performance measured using F1 and accuracy). Experiments demonstrate that manual features outperform all MFCC classifiers, while CNN features outperform both. MFCC-based classifiers suffer an overall error rate increase of up to 96.1% when generalized from English to other languages. Manual features suffer only an up to 35.2% relative error rate increase, and CNN features actually perform the best on Turkish and Spanish, demonstrating that features capable of representing long-term spectral dynamics (CNN and landmark-based features) are able to generalize cross-lingually with little or no loss of accuracy.

1aSC3. Selecting frames for automatic speech recognition based on acoustic landmarks. Di He (Coordinated Sci. Lab., Univ. of Illinois at Urbana-Champaign, 1308 W Main St. Rm. 403, Urbana, IL 61801, dihe2@illinois.edu), Boon Pang P. Lim (Inst. for Infocomm Res. (I2R), Singapore, Singapore), Xuesong Yang (Beckman Inst. for Adv. Sci. and Technol., Univ. of Illinois at Urbana-Champaign, Champaign, IL), Mark Hasegawa-Johnson (Beckman Inst. for Adv. Sci. and Technol., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Deming Chen (Coordinated Sci. Lab., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Most mainstream Mel-frequency cepstral coefficient (MFCC) based Automatic Speech Recognition (ASR) systems consider all feature frames equally important. However, the acoustic landmark theory disagrees with this idea. Acoustic landmark theory exploits the quantal non-linear articulatory-acoustic relationships from human speech perception experiments and provides a theoretical basis of extracting acoustic features in the vicinity of landmark regions where an abrupt change occurs in the spectrum of speech signals. In this work, we conducted experiments, using the TIMIT corpus, on both GMM and DNN based ASR systems and found that frames containing landmarks are more informative than others during the recognition process. We proved that altering the level of emphasis on landmark and non-landmark frames, through re-weighting or removing frame acoustic likelihoods accordingly, can change the phone error rate (PER) of the ASR system in a way dramatically different from making similar changes to random frames. Furthermore, by leveraging the landmark as a heuristic, one of our hybrid DNN frame dropping strategies achieved a PER increment of 0.44% when only scoring less than half, 41.2% to be precise, of the frames. This hybrid strategy out-performs other non-heuristic-based methods and demonstrated the potential of landmarks for computational reduction for ASR.

1aSC4. A flexible discriminative approach to automatic phone and broad phonetic group classification. Kantapon Kaewtip and Abeer Alwan (Elec. Eng., UCLA, 623 1/2 Kelton Ave., Los Angeles, CA 90024, jomjkk@gmail.com)

In this work, we present a novel framework to phone and broad phonetic group (BPG) classification. The overall system adds discriminative power to the traditional HMM framework. All phones share one HMM. However, instead of using generative models (e.g., GMMs), our framework uses a discriminative classifier to predict the state probability (i.e., the probability of an HMM state given a feature vector input). Then, the optimal state sequence is decoded resulting in a time-alignment function between the acoustic feature vector sequence and the state sequence. For each state s, the corresponding feature vectors are averaged resulting in a single feature vector that represents the s-th vector of the block. Each phone class is represented by a block of feature vectors whose size is equal to the number of states. All feature vectors of the block are then concatenated to a single feature vector to represent a phone unit, which is used for a discriminative phone classifier. We validate our framework using the TIMIT database. The proposed framework with MFCCs has comparable performance to the-state-of-the-art phone classification algorithms, but with increased flexibility to account for duration and other features such as articulatory features. Improved performance for BPG classification is also observed.

1aSC5. Exploitation of phased-based features for emotional arousal evaluation from speech. Igor Guoth, Milan Rusko, Marian Ritomský, Trnka Marian, and Sakhia Darjaa (Inst. of Informatics, Slovak Acad. of Sci., Dubravska cesta 9, Bratislava 845 07, Slovakia, igor.guoth@savba.sk)

The mel cepstral coefficients representing magnitude spectrum and Teager energy operators are often used as features in emotion recognition. The phase spectrum information is generally ignored. In this work an approach is proposed based on the use of group delay function from all pole models (APGD) to represent the phase information for the emotional arousal recognition from speech. The experiments were done on the CRISIS acted speech database with four levels of stress. The results of the arousal recognition system using the APGD features are compared to those using mel-frequency cepstral coefficients (MFCCs) and with Critical Band Based TEO Autocorrelation Envelope features (TEO-CB-Auto-Env) which have been successfully used in the task of emotion and stress detection in the past. The feature extraction is applied on the voiced parts of speech. The combination of APGD, MFCC, and TEO-CB-Auto-Env features has shown the best recognition results confirming the hypothesis that the phase and magnitude spectra contain complementary information and their combination can improve the reliability of the arousal recognition system.

1aSC6. Toward real-time physically-based voice synthesis. Zhaoyan Zhang (Dept. of Head and Neck Surgery, Univ. of California, Los Angeles, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

While physically based voice production models have potential applications in clinical intervention of voice disorders and personalized natural speech synthesis, their current use is limited due to the high computational cost associated with simulating the voice production process. In our previous studies [Zhang 2015, J. Acoust. Soc. Am. 137, 898], we have developed a reduced-order voice synthesis program with significantly improved computational efficiency toward real-time applications. One of the simplifications is the use of vocal fold eigenmodes as building blocks to reconstruct more complex vocal fold vibration patterns, which has significantly reduced the computational time, particularly if only a few eigenmodes are used in the simulations. The goal of this study is to identify the minimum number of eigenmodes that need to be included in order to achieve a balance between computational speed and fidelity in voice acoustics and voice quality. The results show that for most voice conditions as few as 30 eigenmodes are sufficient to accurately predict the fundamental frequency, vocal intensity, and selected spectral measures. It is expected that for applications in which absolute values are not as essential, even smaller number of eigenmodes would be acceptable, allowing near real time capability. [Work supported by NIH.]

1aSC7. Robust speaker identification via fusion of subglottal resonances and cepstral features. Jinxi Guo, Ruochen Yang, Abeer Alwan, and Harish Arsikere (Elec. Eng., UCLA, 56-125B Eng. IV Bldg., 420 Westwood Plaza, Los Angeles, CA 90095-1594, lennyguo@g.ucla.edu)

This paper investigates the use of subglottal resonances (SGRs) for noise-robust speaker identification (SID). It is motivated by the speaker specificity and stationarity of subglottal acoustics, and the development of noise-robust SGR estimation algorithms which are reliable at low SNRs for large datasets. A two-stage framework is proposed which combines the SGRs with different cepstral features. The cepstral features are used in the first stage to reduce the number of target speakers for a test utterance, and then SGRs are used as complementary second-stage features to conduct identification. Experiments with the TIMIT and NIST 2008 databases show that SGRs, when used in conjunction with PNCCs and LPCCs, can improve the performance significantly (2-6% absolute accuracy improvement) across all noise conditions in mismatched situations.

1aSC8. Learning acoustic features for English stops with graph-based dimensionality reduction. Patrick Reidy (Callier Ctr. for Commun. Disord., The Univ. of Texas at Dallas, 1966 Inwood Rd., Mailbox B112-CD, Dallas, TX 75235, reidy@utdallas.edu), Mary E. Beckman (Linguist, The Ohio State Univ., Columbus, OH), Jan Edwards (Hearing and Speech Sci., Univ. of Maryland, College Park, MD), Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), and Allison Johnson (Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

This study applies a semi-supervised graph-based dimensionality reduction algorithm (Laplacian Eigenmaps [Belkin & Nyogi, 2002]) to analyze burst spectra from adult productions of English /k/ and /t/. Multitaper spectra calculated over 25-ms windows were passed through a gammatone filter bank, which models the auditory periphery's frequency selectivity and frequency-scale compression. From these psychoacoustic spectra, a graph was constructed: node pairs (two spectra) were connected if they shared a common talker or target word, and connecting edges were weighted by the symmetric Kullback-Leibler divergence between the spectra. This graph's eigenvectors map the spectra into a low-dimensional feature space. Our preliminary experiments with 512 tokens produced by 16 talkers suggest that this algorithm is able to learn a two-dimensional representation of the bursts which reflects well-established articulatory constriction features. The first dimension linearly separated /k/ from /t/ in the back vowel environment, reflecting posterior versus anterior constriction place; the second dimension linearly separated /k/ from /t/ before front vowels, reflecting apical versus dorsal lingual articulator. Experiments are underway to test how well the algorithm generalizes from the training set to handle unseen productions both from the same talkers and from 5 novel talkers.

1aSC9. Vowel synthesis related to equal-amplitude harmonic series in frequency ranges > 1 kHz combined with single harmonics < 1 kHz, and including variation of fundamental frequency. Dieter Maurer and Heidy Suter (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Toni-Areal, Pfingstweidstrasse 96, Zurich 8031, Switzerland, dieter.maurer@zhdk.ch)

Front vowels can be synthesized on the basis of series of harmonics equal in amplitude, with frequencies only above 1 kHz. In these cases, spectral energy usually attributed to the first formant frequency is lacking. The present paper reports results of an experiment in which sound synthesis was performed on the basis of harmonic series covering higher frequency ranges above 1 kHz, combined with a single lower harmonic < 1 kHz, all harmonics equal in amplitude. Thereby, two or three sounds were synthesized for which the higher frequency range and the frequency of the lower harmonic is identical, but the frequency distance of the higher harmonics differs resulting in different perceived pitches of the sounds. Vowel recognition of all sounds was investigated by means of a listening test in which five phonetic expert listeners were asked to assign the synthesized sounds to Standard German vowel qualities. The results of the experiment reveal that the perceived vowel quality of such types of sound pairs or sound triples differs, confirming earlier indications of the spectral envelope being ambiguous with regard to vowel quality. Implications for the acoustics and perception of vowels are discussed.

1aSC10. Influence of noise on the speaker verification in the air traffic control voice communication. Milan Rusko, Trnka Marian, Sakhia Darjaa, Marian Ritomský, and Igor Guoth (Inst. of Informatics of the Slovak Acad. of Sci., Dubravska cesta 9, Bratislava 845 07, Slovakia, milan.rusko@savba.sk)

The voice communication between pilots and the air traffic controllers is vulnerable to various types of attacks. Speaker verification could be used as an add-on security feature; however, there are several factors that make the use of voice biometry in this scenario difficult to apply. These are among others: open set of speakers, very short utterances, speaker noises, signal clipping, foreign accent of non-native speakers, and high content of background and channel noises in the signal. This paper identifies sources of noise in the entire communication channel and analyzes the influence of these noise components of different types and levels on the reliability of the speaker verification. An i-vector based speaker recognizer with PLDA scoring is used for the experiments. Cockpit noises of several aircrafts and limited-band channel noises are simulated by a software noise-generator. The sensitivity of the speaker verification to the noises of different frequency bands is studied in comparison to the long-term speech spectrum and its variability. Possible measures for increasing the noise robustness of the system are discussed.

1aSC11. "Flat" vowel spectra revisited in vowel synthesis. Dieter Maurer and Heidy Suter (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Toni-Areal, Pfingstweidstrasse 96, Zurich 8031, Switzerland, dieter.maurer@zhdk.ch)

Some studies of natural and of synthesized vowel sounds indicate "flat" vowel-related spectral envelopes or envelope parts in terms of vowel-related frequency ranges with harmonics equal in amplitude. The present investigation addresses this question in a vowel synthesis experiment in which sounds related to series of harmonics, multiples of 200 Hz in frequency and equal in amplitude, were created. Thereby, for various frequency ranges, the number of harmonics was increased stepwise from a single lower harmonic to an increasingly broader harmonic series, and, inversely, it was also decreased from a broad series of harmonics to a single higher harmonic. The entire frequency range of investigation was 0.2-4 kHz. Vowel recognition was investigated by means of a listening test in which five phonetic expert listeners were asked to assign the synthesized sounds to Standard German vowel qualities. The results of the experiment reveal that synthesized sounds with frequency bands of series of two or more equal-amplitude harmonics allow for a perceptual differentiation of the Standard German vowels /i-y-e-ɛ-a-ɔ-o/. Methodological issues concerning future investigations as well as implications for the acoustics and perception of vowels are discussed.

1aSC12. Finite element simulation of diphthongs in three-dimensional realistic vocal tracts with flexible walls. Marc Arnela and Oriol Guasch (GTM - Grup de recerca en Tecnologies Mèdia, La Salle, Universitat Ramon Llull, C/ Quatre Camins 30, Barcelona, Catalonia 08022, Spain, marnela@salle.url.edu)

During the production of diphthongs, acoustic waves propagate along a time-varying three-dimensional (3D) vocal tract of complex geometry. The shape of the vocal tract walls does not only change because of the action of the articulators to produce a given sound, but also experience an elastic back reaction to the inner acoustic pressure. In this work the Finite Element Method (FEM) is used to simulate these phenomena. The mixed wave equation for the acoustic pressure and acoustic particle velocity expressed in an Arbitrary Lagrangian-Eulerian (ALE) frame of reference is solved to account for acoustic wave propagation in moving domains. The flexibility of walls is considered by solving a mass-damper-stiffness auxiliary equation for each boundary node. Dynamic vocal tract geometries are generated from the interpolation of static 3D vocal tract geometries of vowels, obtained from Magnetic Resonance Imaging (MRI). Some diphthong sounds are generated as examples

1aSC13. Formant pattern ambiguity of vowel sounds revisited in synthesis: Changing perceptual vowel quality by only changing fundamental frequency. Dieter Maurer (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Toni-Areal, Pfingstweidstrasse 96, Zurich 8031, Switzerland, dieter.maurer@zhdk.ch), Volker Dellwo (Phonet. Lab., Dept. of Comparative Linguist, Univ. of Zurich, Zurich, Switzerland), Heidy Suter (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Zurich, Switzerland), and Thayabaran Kathiresan (Phonet. Lab., Dept. of Comparative Linguist, Univ. of Zurich, Zurich, Zurich, Switzerland)

The influence of varying fundamental frequency on the perception of vowel quality in synthesized vowels was tested in two experiments. In experiment 1, based on investigations of natural Standard German vowel sounds, various model formant patterns $F1$' to $F3$' were created and, for each single pattern, sounds were synthesised on two or three fundamental frequencies (range 200-600 Hz). In experiment 2, corresponding to open-tube resonance characteristics for men, women and children respectively, sounds were synthesised with formant patterns $F1$' to $F5$', formant frequencies being odd multiples of 500, 600, or 700 Hz and fundamental frequencies being 1/3, 1/2 or 1/1 of the first formant frequency. Five phonetic expert listeners identified all synthesised sounds in a multiple-choice identification tasks. The results of both experiments revealed that the perceived vowel quality can be changed systematically by varying fundamental frequency only and that the changes can exceed the perceptual boundaries of two neighboring vowels. Further, sounds related to open-tube resonance patterns are not consistently perceived as neutral schwa vowels when fundamental frequency substantially varies. Thus, the result of both experiments strongly confirm previous claims of formant pattern ambiguity as well as of spectral envelope ambiguity of vowel sounds.

SUNDAY MORNING, 25 JUNE 2017 ROOM 302, 10:35 A.M. TO 12:20 P.M.

Session 1aSP

Signal Processing in Acoustics: Application of Bayesian Methods to Acoustic Model Identification and Classification I

Edmund Sullivan, Cochair
Research, Prometheus, 46 Lawton Brook Lane, Portsmouth, RI 02871

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

Chair's Introduction—10:35

Invited Papers

10:40

1aSP1. Model selection for profile structure in Bayesian geoacoustic inversion. Stan E. Dosso (School of Earth & Ocean Sci, Univ. of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca), Hefeng Dong (Dept. of Electron. Systems, Norwegian Univ. of Sci. and Technol., Trondheim, Norway), and Kenneth Duffaut (Dept. of GeoSci. and Petroleum, Norwegian Univ. of Sci. and Technol., Trondheim, Norway)

This paper considers model selection in Bayesian geoacoustic inversion, specifically, the role of seabed parameterization in resolving geoacoustic profile structure. Bayesian inversion is formulated in terms of the posterior probability density (PPD) over the model parameters which are sampled numerically: Metropolis-Hastings sampling in principal-component space enhanced by parallel tempering is employed here. A key aspect of quantitative geoacoustic inversion is that of parameterizing the seabed model. Trans-dimensional (trans-D) inversion methods model the seabed as a sequence of discontinuous uniform layers and sample probabilistically over the number of layers. However, in some cases it may be expected for seabed properties to vary as smooth, continuous gradients which are not well represented by uniform layers. Most gradient-based inversions assume a representative functional form, such as a power law. A recent alternative is based on a linear combination of Bernstein-polynomial basis functions. This approach is more general and allows the form of the profile to be determined by the data, rather than by a subjective model choice. This paper compares trans-D, power-law, and Bernstein-polynomial inversions for the problem of estimating seabed shear-wave speed profiles from the dispersion of interface waves. Simulations and data from Oslofjorden and/or the North Sea will be considered.

11:00

1aSP2. Tempered particle filters for non-linear model selection and uncertainty quantification of highly informative seabed data. Jan Dettmer (Dept. of GeoSci., Univ. of Calgary, 3800 Finnerty Rd., Victoria, Br. Columbia V8W 3P6, Canada, jan.dettmer@ucalgary.ca) and Stan E. Dosso (Univ. of Victoria, Victoria, BC, Canada)

Knowledge about seabed properties is important for many geoscientific and navy applications, such as sediment transport, sonar performance prediction, and detection of unexploded ordnance. Bayesian model selection and uncertainty estimation have been shown to provide detailed, quantitative seabed knowledge that is valuable for these applications. However, the extreme computational cost limits the utility of Bayesian methods for increasingly common big data sets. Here, we consider geoacoustic reflectivity surveys based on towed source and receiver arrays. Such systems produce thousands of data sets with high information content that require non-linear inversion along tracks many kilometers in length and cannot be analyzed by standard Bayesian sampling. A particle filter that includes reversible jump Markov chain Monte Carlo updates is applied here for efficient posterior probability estimation. Efficiency is improved by likelihood tempering of various particle subsets and including information exchange within the particle cloud. The tempering applies to reversible jump updates and leads to significantly improved exploration of the trans-dimensional seabed model which accounts for changes in the number of sediment layers and their properties along the track. For challenging track sections, where data change

abruptly, the particle cloud is resampled to increase the number of tempered particles. [Work supported by the U.S. Dept. of Defense, through SERDP, and by ONR Ocean Acoustics.]

11:20

1aSP3. Bayes, models, and data. Edmund Sullivan (EJS_Consultants, 46 Lawton Brook Ln., Portsmouth, RI 02871, bewegungslos@fastmail.fm)

The Kalman filter is often described as a Bayesian processor. However, it is more than that. It is also a natural framework for introducing a physical model into an estimation scheme. As, such, it is a processor that can improve an estimation scheme in two distinct ways—by using prior statistics and by introducing a model. It is shown how the prior statistics are implicitly included in the Kalman update equation, and how models can be introduced in two places—the prediction equation and the measurement equation. Assuming linearity and Gaussianity, it is outlined how the Kalman equations evolve from the particle filter upon the assumptions of linearity and Gaussianity. Since the update equation for the particle filter is Bayes' rule, it is clear that the resulting Kalman update equation is also a form of Bayes' rule, thus verifying that the Kalman filter is indeed a Bayesian processor. It is then shown how a model can be introduced, further improving the quality of the estimate. An example is given, based on real data, where the bearing estimate from a short towed array is found for the case of a significant bearing rate.

11:40

1aSP4. Bayesian modal identification with particle filtering for sediment property inversion. Zoi-Heleni Michalopoulou, Andrew Pole (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu), and Nattapol Aunsri (Information Technol., Mae Fah Luang Univ., Chiang Rai, Thailand)

Sequential Bayesian filtering methods have been previously used in dispersion curve tracking for long range sound propagation in the ocean. Modal frequency probability density functions were extracted for sound speed inversion. Here, we calculate modal arrival time densities, instead, and employ them for inversion for sediment sound speed and thickness and water column depth. Bayesian mode identification is performed to this end. We investigate two methods for describing the statistical errors in power spectra, which we use in the arrival time density calculation using normal modes. We then link these densities to the parameters of interest. The approaches are tested with synthetic data as well as data collected in the Gulf of Mexico. [Work supported by ONR.]

12:00

1aSP5. Room acoustic modal analysis via model-based Bayesian inference. Douglas Beaton (TALASKE | Sound Thinking, 1033 South Blvd., Oak Park, IL 60302, douglas@talaske.com) and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

This work illustrates the application of Bayesian Inference in analyzing modal behavior in an experimentally measured room impulse response at a single location. The Prony Model is employed to model the impulse response as a sum of exponentially decaying sinusoids in the time domain. Bayesian model selection is applied to estimate the appropriate number of modes in the model. Bayesian parameter estimation determines the amplitude, decay time, and modal frequency of each mode. The Bayesian analysis is performed using a nested sampling approach to approximate the evidence for each candidate model. Results from the analysis are verified by a Fourier analysis of the experimentally measured data, and also with classical modal theory. Additional experimental measurements are performed to validate individual modal parameter estimates. The likelihood landscape for the selected model is further explored by uniformly sampling near the point of convergence at the end of nested sampling. Animations are used to observe transient behavior of the sample population throughout the analysis.

Session 1aUWa

Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, Structural Acoustics and Vibration, Physical Acoustics and Biomedical Acoustics: Passive Sensing, Monitoring, and Imaging in Wave Physics I

Karim G. Sabra, Cochair
Mechanical Engineering, Georgia Institute of Technology, 771 Ferst Drive, NW, Atlanta, GA 30332-0405

Philippe Roux, Cochair
ISTerre, University of Grenoble, CNRS, 1381 rue de la Piscine, Grenoble 38041, France

Chair's Introduction—10:35

Invited Papers

10:40

1aUWa1. A single-sided representation for passive and active Green's function retrieval, time-reversal acoustics, and holographic imaging. Kees Wapenaar (Delft Univ. of Technol., Stevinweg 1, Delft 2628CN, Netherlands, c.p.a.wapenaar@tudelft.nl)

The homogeneous Green's function, defined as the superposition of the Green's function and its time-reversal, plays an important role in a variety of acoustic applications, such as passive and active acoustic Green's function retrieval, seismic interferometry, time-reversal acoustics, and holographic imaging. An exact representation of the homogeneous Green's function originates from the field of optical holographic imaging (Porter, 1970, JOSA). In this representation, the homogeneous Green's function between two points A and B is expressed as an integral along an arbitrary boundary enclosing A and B. This implies that the Green's function between A and B can be retrieved from measurements carried out at a closed boundary, or, via reciprocity, from passive observations at A and B of the responses to sources on a closed boundary. In practical situations, the closed-boundary integral usually needs to be approximated by an open-boundary integral. This can lead to significant artifacts in the retrieved Green's function. I will discuss a new, single-sided, representation of the homogeneous Green's function, which obviates the need for omnidirectional access. Like the classical closed-boundary representation, this new single-sided representation fully accounts for multiple scattering. I will indicate applications of this new representation in the aforementioned fields.

11:00

1aUWa2. Fluctuations in the cross-correlation for fields lacking full diffusivity: The statistics of spurious features. richard weaver and John Y. Yoritomo (Phys., Univ. of Illinois at Urbana-Champaign, 1110 w green, Urbana, IL 61801, r-weaver@illinois.edu)

Inasmuch as ambient noise fields are often not fully diffuse the question arises as to how, or whether, noise cross-correlations converge to Green's function in practice. Well-known theoretical estimates suggest that the quality of convergence scales with the square root of the product of integration time and bandwidth. However, correlations from natural environments often show random features too large to be consistent with fluctuations from insufficient integration time. Here, it is argued that empirical seismic correlations suffer in practice from spurious arrivals due to scatterers, and not from insufficient integration time. Estimates are sought for differences by considering a related problem consisting of waves from a finite density of point sources. The resulting cross-correlations are analyzed for their mean and variance. The mean is, as expected, Green's function with amplitude dependent on noise strength. The variance is found to have support for all times up to its maximum at the main arrival. The signal-to-noise ratio there scales with the square root of source density. Numerical simulations support the theoretical estimates. The result permits estimates of spurious arrivals' impact on identification of cross-correlations with Green's function and indicates that spurious arrivals may affect estimates of amplitudes, complicating efforts to infer attenuation.

11:20

1aUWa3. Global propagation of seismic body waves and correlation. Michel Campillo, Lise Retailleau, Pierre Boue, Lei Li (ISTerre, Université Grenoble Alpes, ISTerre, UGA Maison des GéoSci., Grenoble 38041, France, michel.campillo@univ-grenoble-alpes.fr), Piero Poli (EAPS, MIT, Cambridge, MA), and Maarten de Hoop (RICE Univ., Houston, TX)

We discuss the nature of retrieved body waves at teleseismic distances from correlation of records in two separate bands T<10 s and T>30 s. The short period correlations indicate the presence of deep phases that appear as correct reconstructions of actual phases. We present an example of application to the reflectivity of the core-mantle boundary region. Careful tests show the reliability of the images produced with ambient noise records. On the opposite, we analyze long period records and show that the correlations are dominated by strong coherent phases (with time close to actual ScS or P'P'df) that are the signatures of high quality factor normal modes. By using array analysis and spectral analysis, we identify the dominant constituents. We then make use of geometrical quantization to derive the ballistic reverberations of rays that contributes to the emergence of signals at times close to body wave arrivals. Our study indicates that the signals measured in the long period correlations have a physical significance, but their interpretation as station to station seismic ray is nontrivial.

11:40

1aUWa4. Passive acoustic remote sensing of the coastal ocean using interferometry of diffuse noise fields. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu) and Michael G. Brown (RSMAS, Univ. of Miami, Miami, FL)

A two-point correlation function of a perfectly diffuse noise field is known to contain all the information about the environment that can be obtained using transceivers placed at the two points, provided that environmental parameters are time-independent. This theoretical prediction underlies the approach to passive remote sensing that is known as noise (or wave) interferometry. However, acoustic noise in the ocean is never perfectly diffuse, except at very high frequencies, where noise of thermal origin dominates. Moreover, the averaging times necessary for deterministic features to emerge from noise cross-correlations far exceed the time scales of temporal variations of the ocean surface, e.g., due to surface gravity waves and in the water column, e.g., due to internal gravity waves and tides. This paper reviews current theoretical understanding of limitations of noise interferometry, which result from time-dependence of environmental parameters and noise anisotropy in the horizontal and vertical planes. It is demonstrated that, within these limitations, phase-coherent data processing techniques, including back-propagation, waveform matching, and time-warping, can be successfully applied to measured noise cross-correlations to characterize seafloor properties and evaluate current velocity in a coastal ocean. [Work supported by NSF and ONR.]

SUNDAY MORNING, 25 JUNE 2017 — ROOM 306, 10:40 A.M. TO 12:00 NOON

Session 1aUWb

Underwater Acoustics: Underwater Acoustic Uncertainty

Andrey K. Morozov, Chair
Teledyne, 49 Edgerton Drive, North Falmouth, MA 02556

Contributed Papers

10:40

1aUWb1. Modeling the shipping noise in uncertain environment for marine space planning. Florent Le Courtois, G. Bazile Kinda, and Yann Stéphan (HOM, Shom, 13, rue du Chatellier BP 30316, Brest 29603, France, florent.le.courtois@shom.fr)

The shipping noise is a major component of the underwater soundscape at low frequency. The increase of the fleet size at world scale for several decades have arisen the potential impacts of noise pollution on marine ecosystems. Modeling the shipping noise levels at large scale became an important task of marine environmental policies. However, because of fluctuating and unknown environmental parameters, the numerical model may present a bias in the estimated levels up to several tens of dB. To tackle this problem, this paper relies on a semi-empirical formulation of the noise level standard deviation, related to environmental mismatch. The model provides monthly noise atlas at world scale, using statistical distribution of shipping. Results are presented using worldwide ship traffic data for the years 2003, 2012, and 2016. It provides a relevant tool to monitor the ambient noise evolution and have been applied for the evaluation of the Marine Strategy Directive Framework.

11:00

1aUWb2. Dynamically orthogonal equations for stochastic underwater sound propagation. Wael Hajj Ali, Johnathan H. Vo, and Pierre F. Lermusiaux (Computation for Design and Optimization Program, Dept. of Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, whajjali@mit.edu)

Grand challenges in ocean acoustic propagation and inference are to accurately capture the dynamic environmental uncertainties and to predict the evolving probability density distribution of stochastic acoustic waves, all efficiently and rigorously, using the governing partial differential equations (PDEs). To start addressing these needs, the stochastic dynamically orthogonal (DO) PDEs for the parabolic wave equation are derived and numerical schemes for their integration are obtained. Within the parabolic approximation, these equations are the optimal reduced-order representation of stochastic acoustic waves within the uncertain sound speed environment. The DO equations govern the propagation of the mean field, the DO modes, and their stochastic coefficients. Examples are provided for a set of idealized test cases as well as for more realistic ocean environments, and predictions are contrasted with those of other uncertainty quantification schemes. The utilization of DO equations for end-to-end uncertainty prediction within oceanographic-seabed-acoustic-sonar dynamical systems is discussed.

11:20

1aUWb3. Normal-mode statistics of sound scattering by a rough elastic boundary in underwater wave-guide in a full coupled mode approach, including back-scattering. Andrey K. Morozov (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com) and John A. Colosi (Dept. of Oceanogr. Graduate School of Eng. and Appl. Sci., Naval Postgrad. School, Monterey, CA)

The underwater sound scattering by rough sea surface, ice, or rough elastic bottom is studied. The effects of scattering from a rough elastic boundary are included in a coupled mode propagation model by the analytical equation for solid-state layer impedance. A full two-way coupled mode solution was used to derive the stochastic differential equations for the second order statistics in a Markov approximation for a transport theory. The coupled mode matrix was approximated by a linear function of one random parameter such as ice-thickness or the surface perturbation. A one parameter Gaussian model has a form of a two-way differential stochastic equation for the correlation of normal mode coefficients. The derived equation relates the correlation matrix of mode coefficients (for both directions) with the correlation function of the rough boundary and the power spectrum of the its slopes. The theory gives a solution for sound attenuation and horizontal coherence over long-range propagation along random interfaces. The result can be used for the estimation of sound attenuation in long-range under-ice propagation or attenuation of seismic waves in an underwater wave-guide with random bathymetry.

11:40

1aUWb4. Sonar inter-ping noise field characterization during cetacean behavioral response studies off Southern California. Shane Guan (Office of Protected Resources, National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20902, shane.guan@noaa.gov), Brandon L. Southall (SEA, Inc., Aptos, CA), Jay Barlow (Southwest Fisheries Sci. Ctr., National Marine Fisheries Service, La Jolla, CA), and Joseph F. Vignola (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC)

Concerns of effects from military mid-frequency active sonar (MFAS) on marine mammals have motivated considerable recent research and technology development. However, robust characterizations of the complex acoustic field during sonar operations have been limited. Additionally, potential effects to marine mammals beyond simple exposure levels are not well understood. Here, we investigate inter-ping reverberation during a behavioral response study with simulated MFAS off California in waters deeper than 300 m using drifting acoustic recording buoys. Acoustic data were collected before, during, and after playbacks of simulated MFAS. An incremental computational method was developed to quantify the inter-ping sound field during MFAS transmissions. Descriptive statistics are used to compare the characteristics of inter-ping sound field and the natural background. Results show significant elevated sound levels within the MFAS frequency band of the inter-ping sound field. In addition, the duration of elevated inter-ping sound field depends on the MFAS source distance. At a distance of 900-1300 m from the source, inter-ping sound field remained 5 dB above natural background levels for approximately 15 s. The elevated inter-ping sound levels at such large distances is most likely due to volume reverberation of the marine environment, although multipath propagation may also contribute to this phenomenon.

Session 1pAAa

Architectural Acoustics and Noise: Noise and Soundscapes in Restaurants and Other Public Accommodations

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

Kenneth P. Roy, Cochair
Building Products Technology Lab, Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603

Chair's Introduction—1:15

Invited Papers

1:20

1pAAa1. Soundscape versus noise—From the sound recording to the public space. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

The enemy of desired sounds is unwanted sonic competition. The positive elements of an interior soundscape need to be rid of noisy rivals. Solutions in the built environment might be inspired by approaches taken in the creation of sound recordings. Of course, noise is minimized where possible, but remaining noise must be dealt with. Masking the noise, taking care that the noise does not mask the soundscape, embracing the noise, and creating an environment of heightened awareness can improve the listener's comfort and sense of pleasure when immersed in a soundscape clouded by noise.

1:40

1pAAa2. The soundscape of dining. Keely Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, ksiebein@siebeinacoustic.com)

This paper explores how applying soundscape theory can address the acoustic concerns for various classes of restaurants; from luxury/fine dining to moderate to fast food. A representative case study of each type of restaurant is examined. By analyzing the soundscape components of each dining space, such as the acoustic community, taxonomy, and itinerary, and the specific paths of communication that take place, one can begin to develop an "acoustic identity" for each room. Each "acoustic identity" is shaped by the users, the aesthetic intent and the soundscape analysis. Diagnostic measurements are made based on how people use the space and the various communication paths present. Impulse responses, alpha bars and other acoustic metrics are used that are based on these communication paths assist in determining basic design approaches and acoustic interventions for each space. By combining the soundscape analysis approach with taking diagnostic measurements rooted in actual communication paths that are present, one can define and shape the acoustical identity for the restaurant.

2:00

1pAAa3. Case study: Renovation of a historical building to create a boutique hotel, restaurant, and pub in a harbor town. Steve Pettyjohn (The Acoust. & Vib. Group, Inc., 5765 9th Ave., Sacramento, CA, spettyjohn@acousticsandvibration.com)

An existing inn and restaurant were to be refurbished to create a boutique hotel, restaurant, and pub in Ft. Bragg, California. Live bands would play in the restaurant while hotel guests would occupy the rooms above the restaurant and pub and at the rear of the pub. The existing structure allowed only a limited number of guest rooms. Other structures were to renovated to provide more lodging space. The proposed uses provide multiple types of and conflicts between the different soundscapes. Sound levels with or without bands must be acceptable in guest rooms. Both wall and floor/ceiling assemblies were designed to meet State Codes and the standard of the industry for sound and impact transmission loss. The soundscape goal for the pub and the restaurant was to control excess sound generated by the patrons. When a band is playing, the potential for significant impacts is greater without acoustical treatment and some control over the band volume. A discussion of the options for acoustical treatments within the restaurant and pub are discussed based on the architectural designs and goals. Alternate methods of controlling sound transmission through wall and floor/ceiling assemblies are presented on the real world conditions and contractors methods.

2:20

1pAAa4. Auralization as a tool for acoustical design of restaurants and public spaces. Kelsey Hochgraf (Acentech, 33 Moulton St., Cambridge, MA 02138, khochgraf@acentech.com)

Auralization is an invaluable decision-making tool for the acoustical design of restaurants and other public gathering spaces, and accurate modeling and calibration of sound sources is critical to achieving perceptually plausible soundscapes of such spaces. Unlike auralizations of performing arts venues, auralizations of restaurants and public gathering spaces afford owners, architects, and consultants the opportunity to directly experience how difficult (or easy) it is to communicate with others when immersed in the soundscape. Also unlike auralizations of performing arts venues, a realistic auralization of a restaurant or public gathering space must account for the Lombard Effect when calibrating source levels of occupant and activity noise. In this presentation, we will briefly review the history and recent improvements of Acentech's 3DListening studio in Cambridge, MA, including Lombard Effect modeling. Three recent case studies will be used to illustrate the unique role of auralization in the architectural design process, including a restaurant, a college pub located underneath residences, and a multi-level collaborative, interdisciplinary work space at an independent school.

Contributed Papers

2:40

1pAAa5. Experimental validation of Bayesian design for broadband multilayered microperforated panel absorbers. Yiqiao Hou, Cameron J. Fackler , and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, hyqjoy@gmail.com)

Single-layer microperforated panel (MPP) absorbers often exhibit limited absorbing bandwidth. A Bayesian inference framework (encompassing both model selection and parameter estimation) has been utilized to design broadband MPP absorbers. In this work, the broadband multilayered MPP absorbers are experimentally validated. In demonstrating ability to meet practical requirements on both high absorption and wide bandwidth, the current investigation uses a sample design scheme in the frequency range from 300 Hz to 2.4 kHz. A minimum requirement for three MPP layers and the relevant three-layer MPP parameters are derived by the two levels of Bayesian inference to meet the design scheme. MPP samples, based on the predicted design scheme, are fabricated and used to conduct normal-incidence sound absorption measurements in an impedance tube. In order to quantify the fabrication tolerance, the measured acoustic data are used to estimate the MPP parameters of the constructed absorber, using an inverse Bayesian inference method. This paper will discuss the initial MPP design, experimental validations of the designed absorption performance, as well as fabrication tolerance estimations of the MPP parameters.

3:00

1pAAa6. Soundscape of washroom equipment and its application. Lucky S. Tsaih and Yosua W. Tedja (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd., Sec. 4, Taipei 10607, Taiwan, akustx@mail.ntust.edu.tw)

The soundscape of the three washrooms at NTUST with equipment such as toilets, urinals, wash basins, showers, hand dryers, and tissue dispensers have been studied. The acoustical attributes of each type of washroom equipment have been measured, recorded and analyzed with LZFmax values for the 12.5 Hz to 20K Hz frequency bands. Despite the intermittent occurrence of equipment sounds, the overall maximum sound pressure level for the full frequency spectrum has been identified as 92 dB/ 83 dBA. It is aligned with a NC 74 curve. Such high transient sounds could disrupt sleeping in adjacent dormitory rooms and possibly reduce the quality of lecturing in adjacent classrooms. Light weight gypsum board and metal stud partitions, concrete masonry unit and concrete are the typical partitions used in washrooms of the residential, healthcare, hospitality, and schools. The transmission loss values of the partitions were calculated with Insul in the initial study. It was found that the majority of partitions studied have sufficient transmission loss values in the 100 Hz and above frequency bands, but the transmission loss will not be sufficient to reduce washroom equipment noise levels in the frequency bands between 50 Hz and 100 Hz.

3:20–3:40 Panel Discussion

Session 1pAAb

Architectural Acoustics: Prediction of Direct and Flanking Airborne and Impact Sound Transmission

Edwin Reynders, Cochair
KU Leuven, Kasteelpark Arenberg 40, Leuven 3001, Belgium

John LoVerde, Cochair
Veneklasen Associates, 1711 16th St., Santa Monica, CA 90404

Jordi Poblet-Puig, Cochair
DECA - LaCàN, Universitat Politècnica de Catalunya, C/Jordi Girona 1-3, campus Nord, B1-206, Barcelona E-08034, Spain

Chair's Introduction—1:15

Contributed Paper

1:20

1pAAb1. Investigation of heavy-soft impact noise transmission in fitness facilities. John LoVerde, David W. Dong, Samantha Rawlings, and Richard Silva (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

Fitness applications within a mixed-use building can disturb other occupants of the building and, as a result, require assessment and mitigation. There are not currently established standards for conducting repeatable measurement of noise or vibration from fitness activities. The focus of recent work in heavy impacts has been weights, as these are frequently used and typically expected to represent the worst-case scenario. However, fitness programs encompass a variety of activities that are dissimilar from heavy weight impacts, such as use of the human body as an impact source (running, jumping, and dropping) and use of soft and limp materials as an impact source (ropes, medicine balls). This paper presents measurement and application of the predictive methods developed for a heavy rigid impact source to a heavy soft impact source for comparison.

Invited Paper

1:40

1pAAb2. Efficient modeling of sound transmission through finite-sized thick and layered wall and floor systems. Carolina Decraene (Civil Eng., KU Leuven, Kasteelpark Arenberg 40, Leuven 3001, Belgium, carolina.decraene@kuleuven.be), Arne Dijckmans (Acoust. Div., Belgian Bldg. Res. Inst., Brussels, Belgium), and Edwin Reynders (Civil Eng., KU Leuven, Leuven, Belgium)

Built-up wall and floor systems such as roof panels, floors with floating screeds, etc., have found widespread application in building construction. Achieving sufficient sound insulation with these systems is challenging because of their relatively low weight and complex vibro-acoustic behavior. A fast and sufficiently accurate acoustic design tool is needed. The semi-analytical transfer matrix method is able to efficiently compute the response of a thick or multilayered structure in the frequency-wavenumber domain but has important limitations. First, the system is assumed to be of infinite extent. At lower frequencies however, neglecting the modal behavior of the wall can lead to large prediction errors. Second, integration over all possible incident plane waves is necessary to obtain the diffuse transmission loss, resulting in a high computation time. The transfer matrix approach is therefore extended in two ways. The modal behavior of rectangular walls and floors with simply supported boundary conditions is approximately accounted for. Using the diffuse reciprocity relationship, a hybrid modal transfer matrix-statistical energy analysis method is then developed such that integration of plane-wave transmission over all angles of incidence is no longer necessary, largely decreasing the computational effort. The model is validated against alternative numerical prediction models and experimental data.

Contributed Papers

2:00

1pAAb3. Acoustics of naturally ventilated double transparent facades. Daniel Urbán (A&Z Acoust. s.r.o., S.H.Vajanského 43, Nové Zámky 94079, Slovakia, ing.daniel.urban@gmail.com), Bert Roozen (Dept. of Phys. and Astronomy, Soft Matter and Biophys., KU Leuven, Leuven, Belgium), Peter Zaťko (A&Z Acoust. s.r.o., Bratislava, Slovakia), Monika Rychtarikova (KU Leuven, Faculty of Architecture, Leuven, Belgium), Peter Tomašovč (Dept. of Bldg. Structures, STU Bratislava, Faculty of Civil Eng., Bratislava, Slovakia), and Christ Glorieux (Dept. of Phys. and Astronomy, Soft Matter and Biophys., KU Leuven, Leuven, Belgium)

This publication presents results of research on naturally ventilated Double Transparent Facades (DTF). The influence of the structural design of DTFs on the airborne sound insulation was investigated. For this purpose, 9 DTFs were measured in situ and 9 Double Transparent Façade Elements (DTF) were measured in a laboratory environment. The influence of the cavity thickness, the parallelism of the constitution layers, the amount of absorbing surfaces in the cavity, and the effect of ventilation slots were investigated. Based on the performed measurements, a prediction model that allows a fast engineering calculation of the sound insulation of DTF's was developed.

2:20

1pAAb4. Methodology for measuring and predicting heavy-weight impact noise transmission. Richard Silva, David W. Dong, and John LoVerde (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, rsilva@veneklasen.com)

Noise and vibration from activity in fitness facilities, in particular dropping of weights, is a common source of disturbance and complaint in the United States of America (USA). Products have been developed to mitigate such impact, but quantitative and comparable data of their effectiveness is lacking. In the United States, there is no standardized method for evaluating the reduction in noise or vibration provided by these products, and also no method for predicting noise and vibration levels in potentially affected spaces. The authors' previous research (Internoise 2015, Noise-Con 2016) developed a preliminary test method to evaluate athletic tile flooring with heavy weight drops. The method is based on the reduction in floor vibration when the products are inserted. The method is analogous to the delta-Ln term in the EN12354 calculation method, except applying to heavy weight sources, and can therefore be used to predict the resulting sound levels in receiving spaces. This paper reports additional measurements on different structural systems to validate the applicability of the method. Various flooring products are compared, and the accuracy and repeatability of the measurement method is evaluated.

Invited Papers

2:40

1pAAb5. The effects of the element damping in sound insulation predictions following EN12354. Eddy Gerretsen (Level Acoust. & Vibrations, De Rondom 10, Eindhoven NL-5612 AP, Netherlands, eddy.gerretsen@planet.nl)

In the prediction of the sound insulation between dwellings in accordance with EN 12354 the damping of the elements in the actual construction is an important aspect. It should be taken into account through the structural reverberation time or, in case of well-damped and generally light weight elements, it is included in the input parameters. As a simplified approach though, the effect of the damping can be neglected. In this respect, several questions are relevant. In what situation should the structural reverberation time in the actual construction be taken into account or when is an element enough damped to make the connections to other elements irrelevant. What relations exists between the junction parameters for damped and reverberant elements. And what errors do we make in neglecting the damping in the actual construction for the elements, for the junctions or for both. Looking deeper into the equations and relations between them some global answers can be given to those questions.

3:00

1pAAb6. On the measurement of the radiation efficiency for the estimate of the resonant sound reduction index. Jeffrey Mahn and Christoph Höller (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1C4N4, Canada, jeffrey.mahn@nrc-cnrc.gc.ca)

The estimate of the resonant sound reduction index has received attention over the years as the prediction method described in the standard, ISO 15712 has been applied to lightweight building constructions. A method of estimating the resonant sound reduction index involves the measurement of the total and the resonant radiation efficiencies of the building elements involved in the first order flanking paths. The radiation efficiencies of different lightweight wall constructions were evaluated as part of a study conducted at the National Research Council Canada are presented. The study focused on the measurement of the radiation efficiencies with the aim of developing guidelines for the measurements. Predicted values of the flanking transmission loss for each flanking path are compared to data which was measured in the National Research Council's eight room flanking facility.

3:20–3:40 Break

Contributed Paper

3:40

1pAAb7. Dilemmas in the assessment of the insulating properties of double massive partition according to EN 12354-1. Dragana Sumarac Pavlovic, Miloš Bjelic, Milos Dinic, Miomir Mijic (School of Elec. Engineer, Univ. of Belgrade, Bulevar kralja Aleksandra 73, Belgrade 11000, Serbia, dsumarac@etf.rs), and Vlada Bezbradica (URSA, Beograd, Belgrade, Serbia)

The introduced requirements for thermal insulation between dwellings, as well as some practical reasons, initiated greater use of double walls in building design practice. The standard EN 12354 -1 did not defined a procedures for calculation of the apparent sound reduction index for such constructions. The paper suggests a possible approach for calculation of apparent sound reduction index based on the calculation methodology for the single partition defined by the standard. Some theoretical analyses of the sound energy transmission through direct and flanking paths in a case of double massive partition were presented. Verification of the proposed approach were done by laboratory measurement and by FEM numerical simulation. The analysis included a number of commonly used constructions and types of their junctions.

Invited Paper

4:00

1pAAb8. The prediction of the flanking transmission in constructions of hollow concrete block masonry walls connected to precast prestressed concrete hollow core floors. Jeffrey Mahn and Christoph Höller (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, jeffrey.mahn@nrc-cnrc.gc.ca)

A common construction technique for multi-story buildings is to build walls of hollow concrete block masonry which are rigidly connected to floors of precast prestressed concrete hollow core slabs. The airborne flanking transmission for buildings of this construction must be determined to predict the apparent sound transmission class to meet the requirements of the National Building Code of Canada. Ideally, this would be done using the prediction method and the vibration reduction index values found in the standard, ISO 15712. However, prior studies conducted at the National Research Council Canada have shown that the hollow core slabs are neither homogeneous nor isotropic which are the requirements for predicting the values of the vibration reduction index (Kij) according to Annex E of the standard ISO 15712. To determine if the theoretical values of the vibration reduction index could nonetheless be applied in practice, an experimental investigation was performed on full scale junctions between concrete block masonry walls and precast concrete hollow core floors built and tested in full compliance with the standard ISO 10848. The investigation found that conservative vibration reduction index values could be predicted using Annex E of ISO 15712.

Contributed Paper

4:20

1pAAb9. Evaluation of mock up testing as a method to predict impact ratings of new hard surface floors in existing buildings. Jennifer Levins, David W. Dong, and John LoVerde (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jlevins@veneklasen.com)

Renovations in multifamily residential buildings often involve the installation of new hard surface flooring. The new flooring must comply with building code minimum ratings. Homeowners Associations may also impose more stringent acoustic criteria. In many cases, the structure of the building is unknown or there are no published tests available for the desired floor-ceiling assembly. One method to evaluate performance of floor-ceiling assemblies is to conduct impact testing on a small sample of the floor assembly in situ. Although this is less costly than testing a fully installed floor, performance variations have been observed between mock up test assemblies and the final installation. This paper will evaluate the accuracy of mock up tests in predicting the impact rating of final floor installations.

Invited Papers

4:40

1pAAb10. Apparent sound insulation in cross-laminated timber buildings. Christoph Hoeller, Jeffrey Mahn (Construction, National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, christoph.hoeller@nrc.ca), and David Quirt (JDQ Acoust., Ottawa, ON, Canada)

With the 2015 National Building Code now in effect in Canada, predicting the apparent sound insulation in buildings from laboratory measurements is becoming increasingly relevant for architects and designers. In North America, the apparent sound insulation is classified in terms of Apparent Sound Transmission Class (ASTC). The ASTC rating includes both the transmission through the separating assembly and the transmission via flanking paths. The National Research Council Canada has published a number of guideline documents that detail the calculation procedure for ASTC and provide the required laboratory data for various construction types. In NRC Research Report RR-335, “Apparent Sound Insulation in Cross-Laminated Timber Buildings” the focus is on buildings which are constructed from cross-laminated timber (CLT) panels. Measurements of the direct sound insulation of CLT panels and the vibration attenuation at their junctions were conducted at the NRC in recent years. The report RR-335 describes how to combine the relevant data to obtain estimates of the apparent transmission loss and the ASTC rating for a given CLT construction. This presentation will present highlights of the report and demonstrate the use of the Simplified Method and the Detailed Method to calculate the ASTC rating.

5:00

1pAAb11. Prediction of vibration reduction index for junctions made of cross laminated timber elements. Jordi Poblet-Puig (DC - LaCàN, Universitat Politècnica de Catalunya, C/Jordi Girona 1-3, campus Nord, B1-206, Barcelona E-08034, Spain, jordi.poblet@upc.edu) and Catherine Guigou-Carter (Ctr. Scientifique et Technique du Bâtiment, Saint Martin d'Hères, France)

The new revision of standard EN 12354-1 will provide some prediction formulas for estimating the vibration reduction index (Kij) of junctions made of cross laminated timber (CLT) elements. These are based on laboratory and in situ measurements. At the same time, new Kij prediction formulas for heavyweight junctions have also been added to this revised standard, in order to include large amount of data generated by means of parametric numerical analysis. The goal of this research is to study if the same philosophy based on numerical models used for the heavyweight junctions can be extended to CLT elements junctions. This step is not direct because the modeling of CLT structures implies a list of non-trivial aspects to take into account: (1) the details of the junction construction (i.e., direction and type of screws, presence of steel angles and plates); (2) the orthotropy on the mechanical properties of CLT panels; and (3) the different range of material properties and how damping must be considered in the numerical model. The contribution presents the advances made in this direction: comparison with available experimental data, analysis of specific aspects of CLT junctions, comparison with the standard formulation, and extension to different junction types.

5:20

1pAAb12. Measurement data for the prediction of the flanking transmission in lightweight building constructions. Jeffrey Mahn and Christoph Höller (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1C4N4, Canada, jeffrey.mahn@nrc-cnrc.gc.ca)

The ISO 15712 series of standards describe a method of predicting the flanking transmission in homogeneous isotropic building constructions. Since the method was first published in 1979, there has been great interest in applying the prediction method to lightweight constructions which are neither isotropic nor homogeneous. The prediction method becomes more complicated for lightweight constructions because the resonant sound reduction indices of the elements must be estimated from measurement data including the sound reduction indices and the resonant and total radiation efficiencies. However, there is the question of which sound reduction index and radiation efficiencies should be used. Should they be for the whole wall or a single panel? Results from a study conducted at the National Research Council Canada are presented. Predicted values of the flanking transmission loss for each flanking path are compared to data which was measured in the National Research Council's eight room flanking facility.

SUNDAY AFTERNOON, 25 JUNE 2017 ROOM 206, 1:20 P.M. TO 5:40 P.M.

Session 1pAAc

Architectural Acoustics: Teaching and Learning in Healthy and Comfortable Classrooms II

Arianna Astolfi, Cochair
Politecnico di Torino, Corso Duca degli Abruzzi, 24, Turin 10124, Italy

Viveka Lyberg-Åhlander, Cochair
Clinical Sciences, Lund, Logopedics, Phoniatrics and Audiolgoy, Lund University, Scania University Hospital, Lund S-221 85, Sweden

David S. Woolworth, Cochair
Oxford Acoustics, 356 CR 102, Oxford, MS 38655

Invited Papers

1:20

1pAAc1. Compiled acoustic and indoor environmental condition data from 220 K-12 classrooms. Laura C. Brill and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, lbrill@huskers.unl.edu)

A team at the University of Nebraska-Lincoln is currently engaged in a comprehensive study of indoor environmental conditions in K-12 classrooms. Information about the indoor air quality, thermal comfort, lighting, and acoustic conditions have been collected from 220 classrooms across five school districts in Nebraska and Iowa. This paper will present an overview of the acoustic results with regards to sound levels and reverberation time as well as how these results vary based on grade level and school district. This paper will also

present an initial overview of the relationship between acoustic metrics and some of the metrics from indoor air quality, thermal comfort, and lighting such as carbon dioxide levels, temperature, and electric illuminance levels. [Work supported by the United States Environmental Protection Agency Grant Number R835633.]

1:40

1pAAc2. Mapping speech transmission index (STI) and background noise in university classrooms. Andrew Hulva, Michael Ermann, Jeffrey Rynes (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu), Randall J. Rehfuss (Architecture + Design, Virginia Tech, Dubllin, VA), Aaron Kanapesky (Architecture + Design, Virginia Tech, Blacksburg, VA), and Alexander Reardon (Eng., Virginia Tech, Blacksburg, VA)

Noise and intelligibility measurements were taken in dozens of classrooms at approximately every-meter resolution and are presented as heat maps. In doing so we hope to determine (1) the largest (and loudest) classrooms that do not require loudspeaker speech amplification and (2) the radius of muddled intelligibility circumscribed around noise sources such as air diffusers and fan coil units.

2:00

1pAAc3. Noisy voice or voice in noise: Evaluation of cognitive load on the speaker, work in progress. Ingrid A. Verduyckt (Medicine, School of SLP and Audiol., Univ. of Montreal, Ingrid Verduyckt, Université de Montréal, cp 6128, sucurcale Ctr. Ville, Montréal, QC H3C3J7, Canada, ingrid.verduyckt@umontreal.ca), Dick Botteldooren, Annelies Vandevelde, and Annelies Bockstael (Univ. of Ghent, Gent, Belgium)

We are comparing the cognitive load induced by various types of noise in the processing of information from speech. We examine if there is a difference in cognitive load between external noise sources (background noise) and internal noise sources (dysphonic voice). Our hypothesis is that noisy voices could be more cognitively demanding than background noise because they are more similar to the target signal spatially and temporally and the perceived link with the target signal is stronger. 60 normal hearing subjects (18-30 years) listen to texts in ten different conditions: (1) Healthy voice in multitalker babble noise, (2-4) Three types of dysphonic voices in background noise, (5-7) Three dysphonic voices in silence, (8) Healthy voice in silence, (9) Healthy voice with omnidirectional noise equivalent to dysphonic noise, (10) Healthy voice with dysphonic noise from same direction. We evaluate cognitive load in 4 ways: (1) during the listening phase by subjects' performance on a secondary graphic task, (2) after the listening phase by subjects' answers to a multi choice questionnaire and (3) by a free recall of the text, (4) by subjects' grading of their listening effort. Methodological considerations will be discussed and preliminary results will be presented.

2:20

1pAAc4. Communication problems among teachers and noise conditions… hearing difficulties also matters!. Lady Catherine Cantor Cutiva, Pasquale Bottalico (Communicative Sci. and Disord., Michigan State Univ., 3207 Trappers Cove Trail, Apartment 2C, Lansing, MI 48910, ladyccantor@gmail.com), Alex Burdorf (Public health, Erasmus Universiteit, Rotterdam, Netherlands), and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Previous studies on the influence of noise in the classroom on the hearing function of teachers have primarily focused either on physical education teachers or music teachers. However, the influence of classroom noise on hearing difficulties among teachers is largely unknown. The aim of this study was to assess the association between classroom noise levels and self-reported classroom acoustics with self-reported hearing impairments among teachers. In 12 public schools in Bogotá, we conducted a cross-sectional study among 621 Colombian teachers at 377 workplaces. Teachers filled out a questionnaire on individual and self-perceived noise conditions inside the classroom along with perception of their hearing impairments. Logistic regression analysis was used to determine associations between background noise levels and self-reported hearing impairment. High noise levels in the surroundings of schools (Odds ratio [OR] 2.15; 95% confidence interval [CI] 1.25-3.68) was associated with self-reported hearing impairments, but self-reported classroom noise showed no association (Odds ratio [OR] 1.34; 95% confidence interval [CI] 0.91-1.99). This study indicates that noise in schools may play a role in self-reported hearing impairments among teachers.

2:40

1pAAc5. Assessment of auralizations of monosyllabic words for hearing impaired students. Konca Saher (Interior Architecture and Environ. Design, Kadir Has Univ., Kadir Has Caddesi, Istanbul 34083, Turkey, konca.saher@khas.edu.tr)

This paper discusses results of a research project, which seeks to develop Turkish speech recognition tests by auralizations for hearing impaired students based on monosyllabically structured words. In the context of this study, two sets of 25-items phonetically balanced monosyllabic Turkish words were recorded in the anechoic chamber of TUBITAK National Metrology Institute in Gebze, Turkey. Each monosyllabic word was recorded through a carrier sentence. After the vocal quality, accent and pronunciation in the recordings were approved by qualified audiologists, auralizations of the recorded sentences were developed in an acoustic simulation software (ODEON v12). Listening tests developed from auralizations in three classroom models with varying reverberation times and signal to noise ratios were presented to ten hearing impaired students. The preliminary results of these listening tests are presented and discussed in this study. This paper also aims to compare the results of listening tests with hearing impaired students with the listening tests made previously with normal hearing students.

3:00

1pAAc6. Listening efficiency in real and simulated university classrooms. Nicola Prodi and Chiara Visentin (Dipartimento di Ingegneria, Università di Ferrara, via Saragat 1, Ferrara 44122, Italy, nicola.prodi@unife.it)

The present study examines how reverberation, background noise level and type, affect both speech reception performance and the perceived effort of students in a university classroom. The classroom has a volume of 198 m^3 and a reverberation time in occupied conditions of 0.6 s, complying with the target value of the DIN 18041 standard. Diagnostic Rhyme Tests (DRT) in the Italian language were

proposed to a group of 26 normal-hearing young adults: half of them native (Italian), the other half non-native (German) speakers. Data on speech intelligibility (SI) and response time (RT) were collected. The two quantities were combined in the joint metric of listening efficiency, effectively describing the interplay of perceptual and cognitive processes in speech reception performance. The experiment took firstly place *in situ*, where the distinct effects a speech-shaped stationary noise and a fluctuating (ICRA) noise with the same short-term STI were determined. Afterwards, acoustic simulations of the classroom setting were carried out, and the resulting binaural impulse responses used for laboratory experiments with headphones. Simulations and in-situ measurements were compared in terms of STI, SI and RT; then, listening tests under controlled conditions were accomplished with a selection of background noise levels and reverberation times.

3:20–3:40 Break

3:40

1pAAc7. Field survey on the sound environment of childcare facilities in Japan: Analysis of sound generation accompanied by children's activities. Saki Noguchi and Kanako Ueno (Meiji Univ., 1-1-1, Higashimita, Kawasaki, Kanagawa 214-8571, Japan, n-saki@akane.waseda.jp)

Today's childcare facilities in Japan have problems in their sound environments such as bustling noises, and it is pointed out as a result of lack of sound absorption and awareness of the sound environment of people in the field. On the other hand, the sound environment is actually different depending on age and type of activity, but detailed examination has not been done. In this presentation, for the purpose of setting up an acoustic environment according to the development of children and the aim of childcare activities, we report the results of surveying sound environments focusing on group size, age, activity space, and activity contents. We investigated the sound environment and childcare activities in five childcare facilities with different facility type, scale, and operation type. It was observed that children's sounds changed with the development of language and behaviors. The characteristics of sound environment differ depending on actual activity situation and spatial property, and special acoustic contrivance was necessary in the free play scene because different activity sounds were mixed there.

4:00

1pAAc8. Trial in a nursery facility for improving the sound environment. Kanako Ueno and Ken Miyatsuka (Meiji Univ., 1-1-1, Higashimita, Kawasaki, Kanagawa 214-8571, Japan, uenok@meiji.ac.jp)

In Japan, most nursery facilities have been built without considering the acoustic requirements; thus, the rooms tend to be reverberant and very noisy from the daily activities of children. The poor sound environment could be harmful not only as a living environment for the children but also as a working environment for the nursery staff. This presentation reports on a case study of a nursery that worked on the reduction of its high sound level. First, the architectural features of the nursery room and the status of the sound environment, which was reported by the nurses' claims and measured to be approximately 80 dB or higher during lunch time, was investigated. Second, to improve the noisy environment, two tasks were carried out. One was the installation of absorbing materials onto the walls and ceilings, which shortened the reverberation time by half. The other was the introduction of management efforts aiming for children to lower the loudness of their voices. These were performed with a demonstration of proper voice levels with animals of different scales and the generation of a music box sound during lunch. The effects of these trials were physically analyzed by sound recordings and subjectively evaluated by nurses.

4:20

1pAAc9. Speakers comfort and voice use in different environments and babble-noise. What are the effects on effort and cognition? Viveka Lyberg-Åhlander, Heike von Lochow, Susanna Whitling (Clinical Sci., Lund, Logopedics, Phoniatrics and Audiolgoy, Lund Univ., Scania University Hospital, Lund S-221 85, Sweden, viveka.lyberg_ahlander@med.lu.se), Jonas Christensson, Erling Nilsson (Ecophon St. Gobain, Hyllinge, Sweden), and Jonas Brunskog (Acoust., Denmark Tech. Univ., Kgs. Lyngby, Denmark)

Teachers often report voice problems related to the occupational environment, and voice problems are more prevalent in teaching than in other occupations. Relationships between objectively measurable acoustical parameters and voice use have been shown. Speakers have been shown to be able to predict the speaker-comfort of an environment. Teachers with voice problems use the room differently than their voice-healthy controls. The aim of this study was to investigate what vocal changes speakers do in different acoustical environments and noise conditions. Nine female speakers, voice patients, and voice-healthy were exposed to four controlled, acoustical "environments" mounted in the same room: 1. stripped; 2. wall- and ceiling mounted absorbents; 3-4 as 2 but with extra ceiling absorbents and in two positions. The speakers were recorded with voice-accumulator and simultaneous voice recordings and spoke freely for 3-5 min in three noise conditions in each setting: silence, classroom noise (60 dBA), and day-care noise (75 dBA). Questionnaires on effort needed were completed by speakers and listeners. There was a co-play between the rooms and the subjectively assessed vocal- and listening effort and also a correlation to cognitive aspects. Listener assessments and the data from the voice accumulator will be presented. This knowledge may contribute to the area of classroom acoustics and speakers' comfort in general.

4:40

1pAAc10. Experimental measurements of word intelligibility of pre-school children under acoustic interferences of reverberation and background noise. Keiji Kawai and Kazunori Harada (Kumamoto Univ., 2-39-1 Kurokami, Kumamoto 860-8555, Japan, kkawai@kumamoto-u.ac.jp)

Child day-care rooms require optimum acoustic condition as children from 0 to 5 years old are supposed to be vulnerable group against interferences with verbal communication by background noise and excessive reverberation. However such interferences on the children seem not to have been examined in the field of architectural acoustics. Thus on-site experiments were carried out to measure word intelligibility of children from 3 to 5 years old. The procedure was like a true or false game. Each of the test words was mixed with two or three levels of pink noise and convolved with the room impulse responses with different reverberation times. The words were

presented to the children and control groups (elementary school pupils and college students) by a loudspeaker in a daycare room. They were asked to judge whether the word was food or not, and to raise their hands holding yes or no signs. The experiment was repeated three times in different daycare centers with revisions of the condition settings. As the results, the correct answer ratios of pre-school children was lower than that of control groups, in general, and particularly, the ratio of 3-years-old children was much lower than that of other groups. Also under low S/N ratio conditions the ratios of all the groups decreased along with long reverberation.

5:00

1pAAc11. Voice production effects due to extreme reverberation times in real rooms. Michael Rollins (Dept. of Phys. and Astronomy, Brigham Young Univ., Cincinnati, OH), Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Mark Berardi, and Eric J. Hunter (Dept. of Comm. Sci. & Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, ejhunter@msu.edu)

Public school teachers have a heightened risk of voice problems. There are many potential causes of this increased vocal risk, including poor room acoustics (e.g., excessively high or low reverberation times). With increased understanding, rooms could be better designed to maintain communication transfer (intelligibility), while mitigating unhealthy vocal effort and, by extension, voice problems. The present study quantified the influence of a wide range of reverberation times (RT20) on vocal production parameters. Thirty-two participants were recorded completing a battery of speech tasks in eight widely ranging conditions within a reverberation chamber. Changes in RT20 had highly correlated effects on several vocal parameters, including smoothed cepstral peak prominence, acoustic vocal quality index (AVQI), and pitch strength. As RT20 increased, vocal parameters tended toward values commonly associated with dysphonic phonation. Additionally, results were gender dependent, with females tending to produce voice with higher vocal effort than males. These findings begin to objectify the effects of room acoustics on vocal accommodations and provide grounds for developing future talker-oriented room acoustical standards.

5:20

1pAAc12. Self-reported voice problems and contributing factors in a Francophone population of professional and student teachers and nurses. Ingrid A. Verduyckt (School of Speech Lang. Pathol. and Audiol., Faculty of Medecine, Univ. of Montreal, Ingrid Verduyckt, Université de Montréal, cp 6128, sucurcale Ctr. Ville, Montréal, QC H3C3J7, Canada, ingrid.verduyckt@umontreal.ca), Amandine Tordeur (Institut Libre Marie Haps en logopédie, Brussels, Belgium), and Laure-Anne Watteau (Brussels, Belgium)

Context: We explored the relation between work related factors, health practices, personality traits and stress-coping strategies and self-reported voice problems in a population of 354 student teachers (ST), 344 professional teachers (PT), 147 student nurses (SN) and 104 professional nurses (PN). Method: An online survey was conducted. Beside an anamnestic questionnaire to collect data about voice problems, work environment, health practices the VHI-10 was used to quantify voice symptoms, the Big Five-10 to explore personality traits, and the WCC-27 to explore stress coping strategies. Results: The prevalence of self-reported voice problems was significantly higher in ST as compared to SN (23% vs 14%, $p = 0.025$) and in PT versus PN (24% vs 12%, $p = 0.08$). VHI scores were significantly higher for subjects self-reporting a voice problem and significantly higher in the professional than the student groups, with PT with self-reported voice problems having the highest scores (Mean = 22.34 SD = 0.635) $p < 0.001$. An ANOVA shows that 55% of the variance of the VHI scores is explained by the status of the subject (ST/PT/SN/PN), ($F(3,945) = 382.156$, $p < 0.001$). A linear regression shows that 43% of the variance of the VHI scores was explained by Amount of private voice use, Conscientiousness, Extraversion and Emotion centered coping scores ($F(4,830) = 159.201$,$p < 0.001$).

Session 1pAB

Animal Bioacoustics: Biosonar

James J. Finneran, Chair
SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152

Contributed Papers

1:20

1pAB1. Effects of prior intense noise exposure on the ability of big brown bats to navigate through clutter. Kelsey N. Hom, James A. Simmons, and Andrea Simmons (Brown Univ., Providence, RI 02912, kelseynhom@gmail.com)

Big brown bats (*Eptesicus fuscus*) emit intense biosonar calls and process returning echoes to forage and guide flight. Sound pressure levels of emissions can exceed 110-120 dB SPL, amplitudes known to produce temporary threshold shifts in other vertebrates. We conducted behavioral experiments to test the limits of bats' ability to navigate after intense noise exposures. Bats were trained to fly through a dense array of chains that produce a pattern of echoes mimicking those they would receive when flying along vegetation. We quantified flight accuracy (10 flights) and changes in the number and temporal patterning of emissions before and after exposure to intense broadband noise (1 hr, 116 dB SPL). Four bats tested 20 min post-exposure maintained flight accuracy and did not alter the temporal patterning of emissions from that observed pre-exposure. In contrast, two of three bats tested 2 min post-exposure initially would not perform the task or made errors in navigation. Temporal patterning of emissions during successful flights did not vary significantly from those measured during pre-exposure flights. These data suggest that prior intense noise exposure affects motivation to fly but not the ability to process returning echoes. [Work supported by ONR and the Capita Foundation.]

1:40

1pAB2. Vocalizing strategies for acoustically jammed conditions for bat echolocation. Hiroshi Riquimaroux (Shandong Univ., 27 Shanda Nanlu, Jinan, Shandong 250100, China, hiroshi_riquimaroux@brown.edu)

Extraction of signal from noise is quite difficult when both signal and noise have the same temporal and spectral characteristics, for example, extraction of speech sounds of a single speaker from background of speech sounds from group of people. However, we can extract speech signal of a particular person from a group of speech noise, called cocktail party effects. Echolocating bats often come across similar situation where their own vocal signals are masked by vocalizations emitted by neighboring bats. Flying horseshoe bats, *Rhinolophus ferrumequinum*, would compensate Doppler-shifted returning echo frequencies to be constant by adjusting frequency of emitted pulses, called Doppler-shift compensation. Amplitudes of the second harmonic are the most intense in echo locating sounds. Once emitted pulses and returning echoes are added, a powerful masker is created. During acoustically jammed conditions, paradoxically bats tend to make their echo reference frequencies even closer. How can they still conduct Doppler-shift compensation? They would amplify the fundamental frequency of FM component. How can they make the fundamental frequency, not the second harmonic, the strongest? The reason why and how they conduct this behavior will be discussed. [Work supported by MEXT Japan and Grant from Shandong University.]

2:00

1pAB3. Neural spike train similarity algorithm detects differences in temporal patterning of bat echolocation call sequences. Alyssa W. Accomando (185 Meeting St., Box GL-N, Providence, RI 02906, alyssa.accomando@nmmpfoundation.org), Carlos Vargas-Irwin, and James A. Simmons (Neurosci., Brown Univ., Providence, RI)

Bats emit echolocation sounds in complex temporal sequences that change to accommodate dynamic surroundings. Efforts to quantify how these patterns change have included analysis of inter-pulse intervals, sonar sound groups, and changes in individual signal parameters. No standardized method has been adopted for quantifying whether sequences of echolocation calls are similar or different beyond these individual dimensions. Here, a new method is presented for assessing the similarity in temporal structure between trains of bat echolocation sounds. The spike-train similarity space (SSIMS) algorithm, originally designed for neural data analysis, was applied to determine which features of the environment influence temporal patterning of echolocation sounds emitted by flying big brown bats (*Eptesicus fuscus*). Using a relational point-process framework, SSIMS was able to discriminate between pulse sequences recorded in different flight environments, as well as to separate flights depending on the bat's expectation of its surroundings based on previous experience.

2:20

1pAB4. Why hipposiderid biosonar is worth studying. Rolf Müller (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Blacksburg, VA 24061, rolf.mueller@vt.edu), Ru Zhang, Liujun Zhang (Shandong Univ. - Virginia Tech Int. Lab., Shandong Univ., Jinan, China), Peiwen Qiu (Mech. Eng., Virginia Tech, Blacksburg, VA), and Xiaoyan Yin (Shandong Univ. - Virginia Tech Int. Lab., Shandong Univ., Jinan, China)

Although the genus Hipposideros contains a diverse set of more than 70 species of echolocating bats, the biosonar system of this group has received far less attention than that of the related horseshoe bats (family Rhinolophidae) which share the same basic cf-fm biosonar. Only a relatively small number of field observations can be found in the literature and even fewer laboratory studies on hipposiderids have been reported. The Shandong University—Virginia Tech International Laboratory has been working with two of the larger hipposiderid species, the great roundleaf bat (Hipposideros armiger) and Pratt's roundleaf bat (Hipposideros pratti) and has conducted biosonar as well as flight experiments with individuals from both species. It was observed that the bats from both species have highly dynamic biosonar systems that employ large non-rigid noseleaf motions as well as large rigid and non-rigid motions of the pinnae. The motions seen in the hipposiderids appear to be relatively larger and more frequent than those in the greater horseshoe bats with which similar experiments were conducted. In addition, the hipposiderids bats tested were found to be maneuverable fliers that should make an excellent model system for the integration of dynamic sonar with a highly capable flight system.

2:40

1pAB5. Design of a dynamic sonar emitter inspired by hipposiderid bats. Luhui Yang, Allison Yu, and Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, 913022794@qq.com)

The noseleaves of Old World leaf-nosed bats (family Hipposideridae) and the related horseshoe bats (Rhinolophidae) are notable for their elaborate static geometries and a conspicuous dynamics in which the noseleaves change their shapes during biosonar pulse emission as a result of muscular actuation. Whereas the noseleaves of horseshoe bats have already been used as an inspiration for dynamic sonar emitter prototypes, the possible functional role of the specific static and dynamic noseleaf features of Old World leaf-nosed bats have yet to be investigated in this manner. To accomplish this, a dynamic emitter based on the time-variant morphology of Pratt's roundleaf bats (Hipposideros pratti) has been designed. The baffle shape was simplified from a tomographic reconstruction of a biological sample. Five shape features (anterior leaf, sella, lancet, and the two nostrils) were preserved in the model. Motions of these parts were derived from three-dimensional reconstructions of landmark points that were placed on the noseleaf of echolocating bats and recorded with a stereo pair of high-speed video cameras. Actuation mechanisms driven by three stepper motors (one for lancet and sella, one for both nostrils, one for the anterior leaf) were implemented to reproduce the dynamic noseleaf motion pattern observed in the bats.

3:00

1pAB6. The relationship between pinna and noseleaf motions in hipposiderid bats. Shuxin Zhang, Liujun Zhang, Ru Zhang (Shandong Univ. - Virginia Tech Int. Lab., Shandong Univ., Shanda South Rd. 27, Jinan, Shandong 250100, China, shuxinsduvt@yahoo.com), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Old World leaf-nosed bats (Hipposideridae) are a family of bat species that use elaborate baffle shapes to diffract the outgoing ultrasonic pulses and the returning echoes. The baffles at both interfaces ("noseleaves" for emission, outer ears for reception) have dynamic geometries that can be changed through muscular actuation. Shape changes in noseleaves and pinnae can both coincide with sound emission and reception respectively, but the relationship between the dynamics in these two structures has yet to be investigated. To study this relationship, a set of no less than 17 landmarks was placed on the noseleaf and one ear of Pratt's roundleaf bats (Hipposideros pratti) to track the dynamic geometry of these structures simultaneously with a high-speed video camera array. The three-dimensional trajectories of the landmark points were reconstructed using stereovision. The results showed strong, systematic relationships between noseleaf and pinna motions that were found to belong to two different qualitative types. In the first type, noseleaf motions and pinna motion did not change direction during the recording period (e.g., one pulse) which resulted in approximately linear relationships between the positions of the landmarks on both structures. In the second type, direction reversals occurred, but coupling between the motions remained evident.

3:20–3:40 Break

3:40

1pAB7. A simulation study of the biosonar information in natural foliage echoes. Chen Ming (Dept. of Mech. Eng., Virginia Tech, 814 Cascade Ct., Blacksburg, VA 24060, cming@vt.edu), Hongxiao Zhu (Dept. of Statistics, Virginia Tech, Blacksburg, VA), and Rolf Müller (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA)

Echolocating bats are capable of accurate navigation in forests at night. To understand how they perceive the vegetation echoes and find passage ways in dense foliage, previous work by the authors has studied a model for homogeneous foliages that distributed leaves uniformly in a domain, simplified individual leaves as circular disks and hence described an entire foliage by only three parameters (mean leaf radius, orientation, and density). To further explore which additional information inhomogeneous structures within a foliage may impart on the echoes, the model was transitioned from uniform leaf distributions to digital trees with branches and leaf clusters using L-systems to mimic the branching patterns of natural trees. When the tree size was small with very few and short child branches, inhomogeneities in the echoes were readily apparent no matter how wide the beam was. If the tree crown was large, featuring many branches and leaf clusters, the amount of inhomogeneity seen in the echoes depended on the relative scale between the tree size and sonar beamwidth. When big trees and relatively narrow sonar beamwidths were paired in the simulation, the echoes were inhomogeneous; as the beam got wider, the generated echoes were getting more homogeneous.

4:00

1pAB8. Dolphin auditory brainstem responses to frequency-modulated "chirps." James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Jason Mulsow, Ryan Jones, Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA), and Robert F. Burkard (Univ. at Buffalo, Buffalo, NY)

Previous studies have demonstrated that increasing-frequency chirp (up-chirp) stimuli can enhance auditory brainstem response (ABR) amplitudes by compensating for temporal dispersion occurring along the cochlear partition. In this study, ABRs were measured in two bottlenose dolphins in response to 5-μs, spectrally "white" clicks, up-chirps, and decreasing-frequency chirps (down-chirps). For all stimuli, bandwidth was constant (10 to 180 kHz) and peak-equivalent sound pressure levels (peSPLs) were 115, 125, or 135 dB re 1 μPa. Chirp durations varied from 125 to 2000 μs. Up-chirps with durations less than ~1000 μs generally increased ABR peak amplitudes compared to clicks with the same peSPL or energy flux spectral density level, while down-chirps with durations above ~250 to 500 μs decreased ABR amplitudes relative to clicks. Increases in ABR amplitude occurred with up-chirps having a broad range of durations. The findings parallel those from human studies and suggest that the use of chirp stimuli may be an effective way to enhance broadband ABR amplitudes in larger marine mammals. [Work supported by US Navy Living Marine Resources Program.]

4:20

1pAB9. Dolphin click-evoked auditory brainstem responses obtained using randomized stimulation and averaging. James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil)

Measurement of auditory brainstem responses (ABRs) using conventional averaging (i.e., constant interstimulus interval, ISI) is limited to stimulus rates low enough to prevent overlapping of the ABRs to successive stimuli. To overcome this limitation, stimuli may be presented at high rates using pseudorandom sequences (e.g., maximum length sequences) or quasi-periodic sequences. However, these methods restrict the available stimulus sequences and require deconvolution to recover the ABR from the overlapping responses. Randomized stimulation and averaging (RSA) is an alternate method for measuring evoked responses at high stimulus rates that allows more control over stimulus jitter, is flexible with respect to sequence parameters, and does not require deconvolution to extract the ABR waveform [Valderrama *et al.* (2012). "Recording of auditory brainstem response at high stimulation rates using randomized stimulation and averaging," J. Acoust. Soc. Am. 132, 3856-3865]. In the RSA method, ABRs are obtained by averaging responses to stimuli with ISIs drawn from a random distribution. In this study, ABRs were measured in three dolphins using conventional averaging and RSA. Results show the RSA method to be effective provided the ISI jitter exceeds ~1-2 ms. [Work supported by the Naval Innovative Science and Engineering (NISE) Program at SSC Pacific.]

4:40

1pAB10. The dynamics of a dolphin's biosonar signals while performing target discrimination tasks. Whitlow Au (Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, wau@hawaii.edu), John Atkins (Ocean Instruments, Auckland, New Zealand), Heidi E. Harley (New College of Florida, Sarasota, FL), Henri Volpilier (Univeristat' Paris-Saclay, Paris, France), and Wendi Fellner (Disney's Epcot's The Seas, Lake Buena Vista, FL)

An experiment was conducted to determine if a blindfolded echolocating dolphin modified its biosonar signals depending on the targets it was investigating in a target shape discrimination task. Biosonar signals were measured with a specially designed bite-plate apparatus with a dowel extending from the bite plate to support the hydrophone. The detected signals were digitized and stored on a modified SoundTrap (Ocean Instrument New Zealand) attached to the dowel. The dolphin engaged in a matching-to-sample task: he first examined a sample target and then swam to a different area of the pool to examine three alternative targets, one of which matched the sample. The animal's task was to point to the matched target. The characteristic of each emitted signal was determined by calculating the peak frequency, center frequency, rms bandwidth, and rms duration. A specific target set was used for each session of 15-18 trials; some sets were unfamiliar to the dolphin. Considerable amount of variations in the signal parameters were observed across trials and sessions, but our statistical analyses suggested the variations were not based on target identity, thereby leading us to conclude that the clicks emitted by the dolphin did not differ with the target set.

5:00

1pAB11. Sonar processing by the spectrogram correlation and transformation model of biosonar. Stephanie Haro (School of Eng., Brown Univ., 69 Brown St., Box 7251, Providence, RI 02912, Stephanie_Haro@brown.edu), James A. Simmons (Neurosci., Brown Univ., Providence, RI), and Jason E. Gaudette (NUWC Newport, Newport, RI)

Echolocating big brown bats emit frequency-modulated (FM) biosonar sounds and perceive target range from echo delays through spectrogram correlation (SC) and target shape from interference nulls in echo spectra through spectrogram transformation (ST). Combined, the SCAT model is a computationally unified auditory description of biosonar as a real-time process. We developed a Matlab implementation of SCAT and tested it with a succession of simulated bat-like FM signals (chirps), each followed by one or more FM echoes that have realistic delay and spectral characteristics. The model simulates neural response latencies in frequency-tuned delay-lines that use coincidence detections for target ranging by SC. For ST, a novel, deconvolution-like network transforms echo spectra into images of the target's glints by detecting coincidences between spikes that represent spectral nulls in parallel channels tuned to null frequencies. Experiments show that dolphins likely separate ST into two operations—MaPS for short glint separations (macro power spectral features, <80 çs) and MiPS for longer separations (micro power spectral features, >80 çs). The ST deconvolution network models MiPS. The highly distributed character of the model favors real-time operation, an important goal for bioinspired sonar development. [Work supported by ONR.]

5:20

1pAB12. Separation of MiPS/MaPS spectrogram transformations in biosonar. Uday Shriram, James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI 02912, uday_shriram@alumni.brown.edu), and Tengiz Zorikov (Georgian Acad. of Sci., Inst. of Cybernetics, Tblisi, GA)

Echolocating bats and dolphins differ in signals, sound emission, and reception pathways, and important aspects of the acoustic medium. They do, however, receive broadcasts and echoes through similar parallel band-pass filters in the cochlea, which impose integration-times of several hundred microseconds. Echoes consisting of several overlapping reflections from target glints (insect body-parts, fish swim bladders) interfere upon reception to form complex echoes with spectral interference patterns that characterize the target's shape. Bats transform these patterns into images that depict the glints themselves along the range axis, a process called Spectrogram Correlation and Transformation (SCAT). Experiments with dolphins suggest two different scales for recognizing shape from echo spectra—macro- *vs* micro-power spectral features for ST (MaPS for short glint separations of <80 çs; MiPS for longer separations >80 çs). We tested this finding in big brown bats trained to distinguish between 2-glint echoes with long, MiPS-like and short MaPS-like spectral features and found that MiPS covers delay separations of about 25-500 çs from the frequency separation of spectral nulls, which have to fit into FM1 for ST to occur. At a deep level, dolphins and bats appear to share a common processing strategy for forming images. [Work supported by ONR.]

Session 1pAO

Acoustical Oceanography: Topics in Acoustical Oceanography

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

Contributed Papers

1:20

1pAO1. Sediment parameter inversions in the East China Sea. Gopu R. Potty, James H Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), Stan E. Dosso (Univ. of Victoria, Victoria, BC, Canada), Julien Bonnel (Ensta Bretagne, Brest cedex 9, France), Jan Dettmer (Dept. of GeoSci., Univ. of Calgary, Victoria, Br. Columbia, Canada), and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas, Austin, TX)

Geoacoustic inversions using wide-band acoustic sources (WBS) deployed during the Asian Seas International Acoustic Experiment (ASIAEX) along a circular path of radius 30 km centered on a vertical hydrophone array was used to construct a pseudo 3D model of the seabed sediments [Potty et al., J. Acoust. Soc. Am. 140, 3065, 2016]. The geoacoustic inversion approach is based on trans-dimensional Bayesian methodology in which the number of sediment layers is included as unknown in addition to the layer parameters. In this study, the inverse problem is recast such that the unknown parameters are sediment parameters such as porosity, permeability, grain size, etc. The compressional and shear wave speeds and attenuation are estimated from these parameters using Biot or similar geoacoustic models. This inversion approach enables direct comparison of the inversion results to ground truth measurements for sediment cores. High resolution time-frequency analysis techniques were applied to extract modal arrival times accurately. One-dimensional (depth-dependent) inversions will be applied along the various acoustic propagation paths to construct a pseudo 3D sediment model using interpolation. [Work supported by the Office of Naval research.]

1:40

1pAO2. Study on parameter correlations in the modal dispersion based geoacoustic inversion. Lin Wan, Mohsen Badiey (Univ. of Delaware, 261 S. College Ave., 104 Robinson Hall, Newark, DE 19716, wan@udel.edu), and David P. Knobles (KSA LLC, Austin, TX)

The dispersion characteristics of acoustic normal modes are applied in the estimation of seabed parameters (e.g., sound speed, density, and layer depth). The modal arrival time difference is utilized to define the objective function, which calculates the difference between the modeled and measured modal arrival times. In this paper, a shallow water experimental dataset shows that the unknown sound speed, density, and layer depth of the sediment may not be successfully estimated by minimizing the modal dispersion-based objective function due to the correlations among them. For a given modal arrival time difference, an increase (decrease) in sound speed of the sediment can be compensated by increasing (decreasing) the density or layer depth of the sediment. Since there is more than one set of parameter values yielding the desired minimum of the objective function, an additional constraint from other objective functions or the knowledge from direct measurements is required. This paper utilizes a second objective function defined by mode shapes in conjunction with the dispersion-based objective function to partially remove the ambiguity of these unknown seabed parameters. [Work supported by ONR.]

2:00

1pAO3. Scattering statistics of glacially quarried rock outcrops: Bayesian inversions for mixture model parameters. Derek R. Olson (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, dro131@psu.edu) and Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

Knowledge of the probability distribution of the scattered amplitude return from the seafloor in reverberation measurements and seafloor sonar images is a prerequisite to designing effective target detection systems and predicting their performance. Previous measurements have revealed that the distribution is often heavier tailed than the Rayleigh distribution, and may be modeled by the K, Weibull, and log-normal distributions, among others. Recent measurements of the scattering statistics from rock seafloors resulted in a bimodal distribution, which is poorly modeled by many commonly used distributions. The rock surfaces were formed from glacial quarrying and exhibit a stepped structure. The observed distribution is hypothesized to result from a mixture, where the scattered field from vertically oriented facets is modeled as a K distribution, and the scattered field due to the horizontally oriented facets is modeled as a Rayleigh distribution. If this hypothesis is true, then roughness parameters may be estimated from scattering data. A Bayesian technique for estimating the distribution of mixture parameters from the probability distribution of the scattered field is presented. This technique, while computationally expensive, reveals the relationship between the mixture model parameters, and can reveal any degeneracies that could lead to problems during inversions.

2:20

1pAO4. Mode coupling and redistribution of the sound intensity over depth at downslope propagation, in the area of thermocline's contact with bottom. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, Mt. Carmel, Haifa 31905, Israel, bkatsnels@univ.haifa.ac.il) and Andrey Lunkov (Marine GeoSci., Univ. of Haifa, Moscow, Russian Federation)

Downslope propagation of the sound signal is studied in a coastal wedge when the temperature profile reveals the strong thermocline. Sound source is placed above the thermocline near the point of thermocline contact with bottom (TC point). Depth dependence of the sound intensity is considered far enough from the coastal line as a function of distance to the source. It is shown that for some position of the source between TC point and the coast, remarkable redistribution of the sound intensity over water depth takes place (sound field is stressed to the bottom). Numerical simulations of this phenomenon are carried out using adiabatic and coupled normal mode decomposition for bottoms with different sound speeds. Results of modeling are compared with the data of experiment in Lake Kinneret, where coastal slope is characterized by variation of depth from the coast to the 40 m at the distance about 8 km and sound speed profile with the thermocline at 20 m depth. Chirp signals with the center frequency 600 Hz were received by the vertical line array (VLA). Vertical distribution of the sound intensity along VLA was measured. Variability in accordance with theoretical model is demonstrated. [Work was supported by ISF.]

2:40

1pAO5. Acoustic mode based internal tide tomography from the PhilSea 2009 experiment. Tarun K. Chandrayadula (Ocean Eng., IIT Madras, 109 B Ocean Eng., IIT Madras, Chennai, Tamil Nadu 600036, India, tkchandr@iitm.ac.in), John A. Colosi (Oceanogr., Naval Postgrad. School, Monterey, CA), Peter F. Worcester, and Matthew Dzieciuch (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA)

The Philippine Sea 2009 experiment transmitted low frequency broadband sound pulses to a mode resolving array located at 185 km. The entire experiment was conducted over a period of one month, during which, most of the transmissions took place every 5 min, and the rest every 3 h. This talk estimates the travel times of the mode pulses and inverts them for the internal tide induced sound speed variability across the propagation path. A first order perturbation theory approach models mode travel times as range-averaged function of the sound speeds. Depending on the propagation distance, the contributions from some internal tide wavelengths are enhanced while others suppressed in the mode travel times. The talk shows the month-long time-series of the mode peak arrival times and estimates a spectrum for the observations. The presentation then derives an expression to relate the spectral components of the mode arrival times in terms of the internal tide spatial wavenumbers. The magnitude of the arrival time spectral components at the tide frequencies are used to invert for the respective internal tides across the propagation path. The tomographic inverse for the internal tides are compared with CTD measurements at the receiver and the source arrays.

3:00

1pAO6. Observations of non-Rayleigh acoustic backscatter and spatial correlation. Chad M. Smith and John R. Preston (Penn State, The Penn State Univ., Appl. Res. Lab., State College, PA 16804, cms561@psu.edu)

Observations of non-Rayleigh acoustic backscatter in data taken during the 2015 Littoral Continuous Active Sonar (LCAS) experiment are discussed using K-distribution shape parameter and comparison with spatial correlation width estimates. Data were collected using the Five Octave Research Array (FORA) cardioid aperture in a towed, roughly monostatic configuration. Several tracks were repeatedly measured using differing signal bands and pulse lengths allowing the use of matched filter envelope and K-distribution statistics to characterize returns in several pulse parameter configurations. Correlation width estimates are compared with non-Rayleigh regions found using shape parameter. Early work uses shallow broadside angles and vessel travel to estimate statistics for large regions providing an efficient search for clutter events. [Work supported by Office of Naval Research.]

3:20–3:40 Break

3:40

1pAO7. Signal characterization using Hidden Markov Models with applications in acoustical oceanography. Michael Taroudakis and Costas Smaragdakis (Inst. of Appl. and Computational Mathematics, Univ. of Crete and FORTH, Voutes University Campus, Heraklion 70013, Greece, taroud@uoc.gr)

The work presents a method for characterizing underwater acoustic signals using a Markov chain, with hidden variables, based on their wavelet transform. Initially, we assign to the signal a Hidden Markov Model (HMM) for which the conditional posterior probability density function seems to be the most representative using an Expectation-Maximization algorithm. Special techniques are applied to avoid over-fitting which in principle is not desirable for the sought applications. The features used for the assignment consist of two dimensional time series obtained by preprocessing of signal's wavelet packet coefficients. Subsequently, we use an approximation of the Kullback Leibler (KL) divergence as a similarity measure among the HMMs. The approximation is obtained by employing Monte-Carlo (MC) techniques simulating the significant sampling from the HMMs posterior distributions. This technique is used in cases where the similarity of two or more signals is to be exploited. These cases include a variety of problems associated with the monitoring of the marine environment using acoustic or seismic signals. The applications to be presented here are referred to problems of geoacoustic inversions (seabed mapping) using simulated acoustic data and seismic monitoring using real data from a terrestrial seismograph to illustrate the various possible applications of the suggested method.

4:00

1pAO8. Head wave inversion technique using the low-frequency sound from a Robinson R44 helicopter. Dieter A. Bevans and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, dbevans@ucsd.edu)

A series of underwater acoustic experiments using a Robinson R44 helicopter and an underwater receiver station has been conducted in shallow (16.5 m) water. The receiver station consisted of an 11-element nested hydrophone array with a 12 m aperture configured as a horizontal line (HLA) 0.5 m above the seabed. A microphone was located immediately above the surface. The main rotor blades of the helicopter produce low-frequency harmonics, the fundamental frequency being ~13 Hz. The tail rotor produces a sequence of harmonics six times higher in frequency. An analytical solution for the horizontal coherence for the head wave has been developed using a 3-layer (atmosphere-ocean-sediment) acoustic propagation model. By comparing the theoretical coherence with the coherence function from the data the sediment sound speed is recovered. The results from the theoretical model and an experiment conducted north of Scripps Pier off the coast of southern California are presented. [Research supported by ONR, SMART(DOD), NAVAIR, and SIO.]

4:20

1pAO9. Acoustic quantification of abundance, biomass, and size class of Atlantic menhaden (*Brevoortia tyrannus*) in a shallow estuary in Long Island, New York. Brandyn M. Lucca, Hannah Blair (School of Marine and Atmospheric Sci., Stony Brook Univ., 70 Lakewood Court, Apt. 16, Moriches, NY 11955, brandyn.lucca@stonybrook.edu), and Joseph Warren (School of Marine and Atmospheric Sci., Stony Brook Univ., Southampton, NY)

Atlantic menhaden (*Brevoortia tyrannus*) is an euryhaline forage fish commonly found on Long Island from late spring to fall and is both ecologically and economically important to fisheries along the entire eastern coast of the United States. Schools of menhaden frequently occupy the shallow (<4 m) bays and rivers of the western Peconic Estuary on Long Island, New York. These shallow habitats are difficult to sample using traditional pelagic (i.e., net trawls) or shore-adjacent (i.e., beach seines) methods due to shallow water depths and salt marsh coastline. We conducted multiple acoustic surveys using fisheries echosounders (38, 120, and 200 kHz) and sidescan sonar between May and November in 2015 and 2016 in Flanders Bay and the Peconic River. Active acoustic surveys were capable of providing estimates of abundance, biomass, and length distribution of Atlantic menhaden. Abundance and biomass estimates for the entire western Peconic Estuary were extrapolated from survey data and showed large (order of magnitude) variations in the menhaden population in these waters from spring to fall. Length distributions of menhaden differed both among seasons and between years. Data were ground-truthed using video, photographs, and morphological measurements of Atlantic menhaden.

4:40

1pAO10. Profiling measurement of internal tide in the Bali Strait by reciprocal sound transmission. Fadli Syamsudin (Technol. for Regional Resource Development, Agency for the Assessment and Application of Technol. (BPPT), Jakarta, DKI Jakarta, Indonesia), Minmo Chen, Arata Kaneko (Graduate School of Eng., Hiroshima Univ., 1-4-1 kagamiyama, Higashi-Hiroshima, Hiroshima 739-8527, Japan, d153155@hiroshima-u.ac.jp), John C. Wells (Civil Eng., Ritsumeikan Univ., Kusatsu, Shiga, Japan), and Xiao-Hua Zhu (Oceanogr., Second Inst. of Oceanogr., Hangzhou, China)

A reciprocal sound transmission experiment was carried out from 10 to 12 June 2015 along one-crossed-strait line in the Bali Strait with strong tidal current to measure the vertical section structures of range-averaged current and temperature at a 3 minutes interval. The five-layer structures of those parameters in the vertical sections were reconstructed by the regularized of

travel time data for 2 rays. The hourly mean current showed a generation of nonlinear internal tide with amplitudes of (1.0—1.5)m/s and period of 6 hours, superimposed on semi-diurnal internal tide with amplitudes, decreasing from the upper to lower layer. The hourly mean temperature was characterized by variation with amplitude of (1.0-1.5) and period of 6 and 8 hours. Current variation revealed an out of phase relation between the upper and lower layer while temperature varied in phase for all five layers. The 2-day average current formed a stratified structure, varying from -0.6 to -0.1m/s and from 23.8 to 28, respectively. The five-layer current and temperature were significantly over the inversion errors. It is suggested that thermal stratification in the Bali strait was caused by intrusion of dense cold water from the Indian Ocean due to coastal upwelling.

SUNDAY AFTERNOON, 25 JUNE 2017 BALLROOM B, 1:20 P.M. TO 5:00 P.M.

Session 1pBAa

Biomedical Acoustics: Beamforming and Image Guided Therapy II: Cavitation Nuclei

Costas Arvanitis, Cochair
Mechanical Engineering and Biomedical Engineering, Georgia Institute of Technology, 901 Atlantic Dr. NW, Room 4100Q, Atlanta, GA 30318

Constantin Coussios, Cochair
Institute of Biomedical Engineering, Department of Engineering Science, University of Oxford, Old Road Campus Research Building, Oxford OX3 7DQ, United Kingdom

Invited Paper

1:20

1pBAa1. Transcranial acoustic imaging for real-time control of ultrasound-mediated blood-brain barrier opening using a clinical-scale prototype system. Ryan M. Jones, Meaghan A. O'Reilly (Medical Biophys., Univ. of Toronto, 2075 Bayview Ave., Focused Ultrasound Lab (C713), Toronto, ON M4N 3M5, Canada, rmjones@sri.utoronto.ca), Lulu Deng, Kogee Leung (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada), and Kullervo Hynynen (Medical Biophys., Univ. of Toronto, Toronto, ON, Canada)

Multichannel beamforming of passively detected ultrasound (US)-stimulated acoustic emissions is a promising method for guiding cavitation-mediated therapies. In the context of brain applications, our group and others have previously demonstrated the use of conventional beamforming techniques to transcranially map cavitation activity during microbubble (MB)-mediated blood-brain barrier (BBB) opening. MB activity can be mapped at pressure levels below the BBB opening threshold, allowing target confirmation prior to therapy delivery. By including skull-specific phase and amplitude corrections in the reconstruction process, the aberrating effects of the cranial bone can be compensated for to improve image quality. Recently, we have designed, fabricated, and characterized multi-frequency, transmit/receive, sparse hemispherical phased arrays for MB-mediated brain therapy and simultaneous cavitation mapping [Deng *et al.*, *Phys. Med. Biol.* **61**, 8476-8501 (2016)]. This talk will review our progress to date in using these prototype systems to exploit the spatial information obtained from receive beamforming to actively modulate the therapeutic exposures during US-induced BBB opening, following our previously developed single-element internal calibration approach [O'Reilly & Hynynen, *Radiology* **263**, 96-106 (2012)]. We anticipate that this technique will improve the safety and efficacy of MB-mediated BBB opening, as well as other future non-thermal US brain treatments such as cavitation-enhanced ablation, sonothrombolysis, and histotripsy.

Contributed Papers

1:40

1pBAa2. Towards transcranial focused ultrasound treatment planning: A technique for reduction of outer skull and skull base heating in transcranial focused ultrasound. Alec Hughes (Dept. of Medical Biophys., Univ. of Toronto, 101 College St., Rm. 15-701, Toronto, ON M5G 1L7, Canada, ahughes@sri.utoronto.ca), Yuexi Huang, and Kullervo Hynynen (Sunnybrook Res. Inst., Toronto, ON, Canada)

Transcranial focused ultrasound is a rapidly-growing therapeutic modality with expanding applications in the treatment of brain disorders and diseases. As more treatments are proposed by clinicians, the need for comprehensive, accurate treatment planning is required to take into account the complexities that can arise from the application of ultrasound to the brain. These include skull heating, both on the outer surface of the skull and at any bone at the skull base, as well as phasing corrections for skull aberration corrections. We will present a method for the reduction of outer skull and skull base heating by using phased array controls. First, full-wave numerical simulations are used to demonstrate the corrections on a clinically relevant skull base target using exported clinical imaging data. Then, results from ex vivo experiments are presented to illustrate the application of these phased array controls in the reduction of skull heating by scanning a 3D volume of heating around the focus, while computing the corrections in a clinically relevant timescale. Potential limitations of the method and future directions will also be discussed.

2:00

1pBAa3. Passive localization and classification of cavitation activity using group sparsity. Can Baris TOP, Alper Gungor, and H. Emre Guven (Adv. Sensing Res. Program Dept., Aselsan A.S., Mehmet Akif Ersoy Mah., 296. Cad. No :16 Yenimahalle, Ankara 06370, Turkey, cbtop@aselsan.com.tr)

In therapeutic high intensity focused ultrasound (HIFU) applications, cavitation mapping is a powerful tool to monitor and guide the treatment procedure. Furthermore, the frequency spectrum of the cavitation activity can be used to classify the mode of cavitation (stable/inertial), enabling a means for increasing the safety of the application. In this study, we formulate the cavitation mapping as a group sparse constrained optimization problem, minimizing $l_{2,1}$-norm of the solution. The frequency bins related to a class of cavitation activity (harmonic, ultra-harmonic, or broadband) are grouped using l_2-norm for each voxel, and l_1-norm of the image is minimized. We solve this problem using an Augmented Lagrangian Method, specifically the Alternating Direction Method of Multipliers (ADMM). We used a simulation model to test this method on a 300 mm diameter 128-element hemispherical receiver array application. We calculate the radiated pressure from the microbubbles inside the HIFU beam using a rigid vessel bubble dynamics model. Then, we reconstruct the image associated with the bubble activity at the focal region using the received signals for various focal pressure distribution scenarios. The results show that the proposed method provide improved resolution and sensitivity, especially for localizing inertial cavitation activity.

2:20

1pBAa4. Optimizing passive cavitation mapping by refined minimum variance-based beamforming method: Performance evaluations in macaque models. Tao Sun, Calum Crake (Radiology, Brigham and Women's Hospital; Harvard Med. School, 221 Longwood Ave., EBRC 514, Focused Ultrasound Lab., Boston, MA 02115, taosun@bwh.harvard.edu), Brian H. Tracey (ECE, Tufts Univ., Medford, MA), Costas Arvanitis (Radiology, Brigham and Women's Hospital; Harvard Med. School, Boston, MA), Eric Miller (ECE, Tufts Univ., Medford, MA), and Nathan McDannold (Radiology, Brigham and Women's Hospital; Harvard Med. School, Boston, MA)

Microbubble-mediated focused ultrasound (FUS) therapies harness mechanical and/or thermal effects to deliver drugs or ablate tissues. Passive acoustic mapping (PAM) enables the spatio-temporal monitoring of cavitation activity, which is critical for the clinical translation of this technique. Traditional PAM is based on delay-and-sum (DAS) beamforming, a method whose quality tends to deteriorate due to issues including multi-bubble interference, distortion in the wavefront caused by the presence of the skull, unmodeled variability of array elements, etc. To provide for robustness, here we consider the use of minimum variance adaptive beamforming to PAM and demonstrate significant improvement in image quality compared to DAS. The minimum variance distortionless response (MVDR) method was evaluated and further improved by adding diagonal loading and by using subarray covariance estimates. Results demonstrate improvements in both the resolution and image contrast compared to DAS using either traditional or a refined MVDR beamformer. The axial full width at half maximum of the microbubble activity at the focus was reduced to 79.5% and 38.5% of that in DAS image for traditional and refined MVDR beamformers, respectively. Moreover, the refined MVDR method greatly enhanced the robustness while traditional MVDR beamforming induced self-nulling effects. We anticipate that the proposed method will improve our ability to monitor and control FUS-induced cavitation-based therapies.

2:40

1pBAa5. A dual mode hemispherical sparse array for B-mode skull localization and passive acoustic mapping within a clinical MRI guided focused ultrasound platform. Calum Crake, Spencer Brinker, and Nathan McDannold (Radiology, Brigham and Women's Hospital, Harvard Med. School, 221 Longwood Ave., Boston, MA 02115, crake@bwh.harvard.edu)

Previous work has demonstrated that passive acoustic imaging may be used alongside MRI for monitoring of focused ultrasound therapy. However, current implementations have generally made use of either linear arrays originally designed for diagnostic imaging or custom narrowband arrays specific to in-house therapeutic transducer designs, neither of which is fully compatible with clinical MR-guided focused ultrasound devices. Here we have designed an array which is suitable for use within an FDA-approved MR-guided transcranial focused ultrasound device, within the bore of a 3 Tesla clinical MRI scanner. The array is constructed from 5×0.4mm piezoceramic disc elements arranged in pseudorandom fashion on a low profile laser-cut acrylic frame designed to fit between the therapeutic elements of a 230 kHz InSightec ExAblate 4000 transducer. By exploiting thickness and radial resonance modes of the piezo discs the array is capable of both B-mode imaging at 5 MHz for skull localization, as well as passive reception at the second harmonic of the therapy array for mapping of acoustic sources such as emissions from cavitation. The strengths and limitations of the system for passive acoustic imaging during *in vivo* experiments will be discussed, utilizing robust and conventional time and frequency domain beamforming methods.

3:00

1pBAa6. Transcranial histotripsy acoustic-backscatter localization and aberration correction for volume treatments. Jonathan R. Sukovich, Zhen Xu, Timothy L. Hall, Jonathan J. Macoskey, and Charles A. Cain (Biomedical Eng., Univ. of Michigan, 1410 Traver Rd., Ann Arbor, MI 48105, jsukes@umich.edu)

Here, we present results from experiments using histotripsy pulses backscattered off of therapy-generated bubble clouds to perform point-by-point aberration correction and bubble cloud localization transcranially over large steering ranges to demonstrate the efficacy of these methods at improving treatment efficiency and mapping volumetric treatments. Histotripsy pulses were delivered through an ex vivo human skullcap mounted centrally within a 500 kHz, 256-element histotripsy transducer with transmit-receive capable elements. Electronic focal steering was used to steer the therapy focus through individual points spanning a 30 mm diameter volume centered about the transducer's geometric focus. Backscatter signals from the generated bubble clouds were collected using array elements as receivers. Separate algorithms, based on time-domain information extracted from the collected signals, were used to perform aberration correction and localize the generated bubble clouds, respectively. The effectiveness of the aberration correction and localization results were assessed via comparison to hydrophone measurements of the focal pressure amplitude and location taken before and after backscatter aberration correction and localization were applied. Backscatter aberration correction results showed increased focal pressure amplitudes at all steering locations tested. Localization results were in good agreement with hydrophone measurements, but were seen to display preferential bias in the pre-focal direction at larger steering distances.

3:20–3:40 Break

Invited Paper

3:40

1pBAa7. Cavitation enhanced drug delivery *in-vivo* using combined B-mode guidance and real-time passive acoustic mapping: Challenges and results. Christian Coviello, Rachel Myers, Edward Jackson (OxSonics, Ltd., The Magdalen Ctr., Robert Robinson Ave., Oxford OX4 4GA, United Kingdom, christian.coviello@oxsonics.com), Erasmia Lyka (Univ. of Oxford, Oxford, United Kingdom), Lauren Morris, Cliff Rowe, James J. Kwan (OxSonics, Ltd., Oxford, Oxfordshire, United Kingdom), Robert Carlisle, and Constantin Coussios (Univ. of Oxford, Oxford, United Kingdom)

Inertial cavitation nucleated by nano-scale sonosensitive particles (SSPs) at modest peak negative pressures (~1 MPa at 500 kHz) and monitored by passive acoustic mapping (PAM) has been recently shown to improve the dose and distribution of anti-cancer agents during ultrasound (US) enhanced delivery (Myers 2016, Kwan 2015). As applications of therapy monitoring using PAM have advanced rapidly including its use in clinical trials, means of validating the performance of PAM *in-vivo* remains a major focus of efforts. For drug delivery, PAM should not only quickly and reliably detect and localize desired and undesired cavitation, but it should provide some predictor of successful delivery. *In-vivo* experiments using PAM in subcutaneous tumor implanted murine models across a range of cancer cell lines (HEPG2, SKOV, EMT6, CT-26) demonstrate the detection of inertial cavitation by SSPs in the target regions when sonicated by US, but no cavitation with US alone. Additionally when SSPs are co-administered with an oncolytic virus (vaccinia), a small molecule chemotherapeutic (doxorubicin), or an immunotherapeutic (anti-PD-L1 antibody), PAM is able to effectively predict successful delivery in the presence of cavitation in the target regions and unsuccessful delivery in the absence of cavitation in target regions. Enhancements to PAM to deal with artifacts, spurious reflections, and to increase resolution were also able to improve the monitoring capability. Future work focuses on clinical translation and improved validation methods.

Contributed Papers

4:00

1pBAa8. Doppler passive acoustic mapping for monitoring microbubble velocities in ultrasound therapy. Antonios Pouliopoulos, Cameron Smith, Ahmed El Ghamrawy, Mengxing Tang, and James Choi (BioEng., Imperial College London, Royal School of Mines, Imperial College London, South Kensington Campus, London SW7 2AZ, United Kingdom, a.pouliopoulos13@imperial.ac.uk)

The success of microbubble-mediated ultrasound treatments, such as blood-brain barrier disruption and sonothrombolysis, is determined by whether the correct cavitation dynamics are produced at the correct locations. Passive acoustic mapping (PAM) can track the location, magnitude, type, and duration of microbubble-seeded cavitation produced during sonication. Using a single element passive cavitation detector (PCD), we recently showed that microbubble velocities within the PCD listening volume can be determined by analysing the Doppler shifts in the microbubble acoustic emissions. Here, we developed a PAM-based algorithm to passively track microbubble velocities using a linear array. Microbubbles embedded within a vessel were sonicated using a 1 MHz focused ultrasound transducer (pulse length: 50 ms, peak-negative pressure: 200-600 kPa). Acoustic emissions were captured by a co-aligned L7-4 linear array. PAM using Capon beamforming was used to localize the acoustic emissions. We spectrally analyzed the time traces in order to derive position-dependent Doppler shifts and estimate axial velocities at each location. Doppler PAM imaged the axial microbubble velocities along the ultrasound propagation direction, at different time points during sonication. Microbubbles moved at peak velocities of 1-2 m/s due to acoustic radiation forces, producing a time dependent velocity profile. Doppler PAM allowed estimation of microbubble translation within an imaging plane, enabling enhanced monitoring of therapeutic ultrasound applications.

4:20

1pBAa9. Passive acoustic mapping of extravasation for vascular permeability assessment. Catherine Paverd, Erasmia Lyka, Delphine Elbes, and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. Campus Res. Bldg., Headington, Oxford OX3 7DQ, United Kingdom, catherine.paverd@eng.ox.ac.uk)

Prior research has demonstrated that Passive Acoustic Mapping (PAM) enables real-time monitoring of cavitation activity occurring within the vasculature to achieve drug delivery and/or opening of the Blood Brain Barrier. In the present work, we focus on whether sub-micron cavitation nuclei can be imaged once extravasated. This would provide a means of determining both vascular permeability before or after ultrasound exposure, and real-time monitoring of successful drug delivery. A key challenge in achieving these objectives is the spatial resolution of PAM. A novel bistatic setup was used to achieve sub-millimetre resolution both axially and transversely to two imaging arrays. A vertically oriented flow channel in a tissue mimicking phantom was placed at the focus of two perpendicular confocal HIFU transducers, each with a coaxial linear imaging array. Sequential acoustic excitation at 0.5 or 1.55 MHz was used to first extravasate, and then re-excite the nuclei once extravasated. The lower frequency creates stronger microstreaming while the higher frequency favors acoustic radiation force. Results have demonstrated accurate localization of extravasated nuclei at 0.4 mm from the channel. Localization was achieved using optimal beamforming PAM and verified with fluorescence microscopy. Future work will focus on *in vivo* applicability using murine or cancerous perfused organ models.

4:40

1pBAa10. Passive microbubble imaging with short pulses of focused ultrasound and absolute time-of-flight information. Mark T. Burgess, Iason Apostolakis, and Elisa Konofagou (Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians and Surgeons 19-418, New York, NY 10032, mark.t.b42@gmail.com)

Focused ultrasound (FUS)-stimulated microbubble activity has been proposed as an efficient technique in numerous therapeutic ultrasound applications. Passive imaging of microbubble activity is used to spatially map the intensity and location of microbubble activity for correlation with therapeutic outcomes. Current passive imaging methods were developed for application with continuous-wave FUS therapies and have inherent limitations including poor axial image resolution. This study seeks to implement a synchronous passive microbubble imaging method using short pulses of FUS (200-500 kPa peak negative pressures, 2-3 cycles) at high frame rates (500-5000 Hz pulse repetition rate) to preserve absolute time-of-flight and improve axial resolution. *In vitro* and *in vivo* studies were carried out using an 18-MHz imaging array (L22-14v LF, Verasonics, Inc.) and 1-MHz FUS transducer aligned off-axis relative to the imaging array. A research-based ultrasound system (Vantage 256, Verasonics, Inc.) was used for custom transmit and receive sequences. Results indicate that this technique is able to "localize" microbubbles with improved resolution compared to previous methods and create detailed microvascular maps of microbubble activity throughout the focal area. The application of this technique for monitoring FUS-mediated blood-brain barrier opening will be shown. [Work supported in part by NIH grants R01AG038961 and R01EB009041.]

Session 1pBAb

Biomedical Acoustics: Imaging II

Parag V. Chitnis, Chair
Department of Bioengineering, George Mason University, 4400 University Drive, 1G5, Fairfax, VA 22032

Contributed Papers

1:40

1pBAb1. Ex vivo testing of basal cell carcinomas and melanomas with high-frequency ultrasound. Christine E. Dalton, Zachary A. Coffman (Biology, Utah Valley Univ., 800 W University Pkwy, Orem, UT 84058, christine.e.dalton@gmail.com), Garrett Wagner (Comput. Sci., Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

The objective of this study is to significantly reduce the length of surgery for skin cancer patients by developing a diagnostic method to quickly distinguish cancerous from non-cancerous tissue. A common treatment for basal cell carcinoma and several melanomas is Mohs surgery, consisting of surgical resection of the tumor and successive resections of the surrounding tissues (margins). Because each excised specimen needs to be examined for cancerous margins, skin cancer surgery can last up to 4 hours. To rapidly evaluate Mohs surgical specimens, a high-frequency (20-80 MHz) ultrasound method, originally developed for testing breast cancer surgical specimens, was modified for smaller skin cancer tissues. The method uses a narrow-beam (1.5-mm diameter) probe, a broad-beam (6.35-mm diameter) transducer, an ultrasonic pulser-receiver, a digital oscilloscope, an aluminum test stage to hold the specimen, and a hybrid water immersion/contact approach to acquire highly accurate data. The method is currently undergoing a feasibility study on skin cancer surgical margins at the Huntsman Cancer Institute, Salt Lake City, Utah. Preliminary results from 16 patients show that the 20-80 MHz peak density values from the power spectra are consistent with those found in previous breast cancer margin studies.

2:00

1pBAb2. Ultrasonic characterization of human colon carcinoma cells in the 5-25 MHz frequency range. Amy Longstreth, Judene Thomas, Yaa Kwakwa, Janae Davis, and Maria-Teresa Herd (Phys., Mount Holyoke College, 50 College St., South Hadley, MA 01075, longs22a@mtholyoke.edu)

Early recognition of cancerous tissue is crucial in receiving a favorable prognosis. Diagnostic tools that allow for an understanding of differences in the ultrasonic characteristics (such as speed of sound (SOS), attenuation, and backscatter coefficients (BSC)) in malignant and benign cells can aid in early diagnostics. Although statistically significant distinctions between benign and cancerous tumor scatterer properties have been demonstrated, there is little knowledge about which cell characteristics create differences in scattering. This study centers on techniques using quantitative ultrasound to quantify the microstructure of HTC (colon cancer) cells in an attempt to establish a greater understanding of scattering mechanisms. To analyze these characteristics HTC cells were cultured, suspended in agar and prepared into samples. Broadband BSC measurements were conducted using focused transducers and narrowband attenuation and SOS measurements were performed using receiving and transmitting transducers. All experiments were made in the 5-25 MHz range at 21°C. A comparison of the obtained results was made with a similar study using a higher concentration of HTC cells to test the ability to accurately estimate the ultrasonic properties. The results introduce relevant data useful for comparative studies and further analysis.

2:20

1pBAb3. Functional neuro-imaging with magnetic resonance elastography. Samuel Patz (Radiology, Brigham & Women's Hospital, 221 Longwood Ave., Boston, MA 02115, patz@bwh.harvard.edu), Navid Nazari (Biomedical Eng., Boston Univ., Boston, MA), Katharina Schregel, Miklos Palotai (Radiology, Brigham & Women's Hospital, Boston, MA), Paul E. Barbone (Mech. Eng., Boston Univ., Boston, MA), and Ralph Sinkus (Biomedical Eng., Kings College London, London, United Kingdom)

Evaluate changes in the shear modulus of brain tissue as a new measure of localized brain function. A spin-echo magnetic resonance elastography (MRE) sequence was modified to allow two interleaved paradigms: stimulus ON/OFF. To avoid neuronal habituation, a paradigm was active for 9s before switching to the other paradigm. After each paradigm switch, a period of 1.8 s was allowed for hemodynamic equilibrium. Seven healthy black mice were studied. An electrical current to the hind limb, ~1 mA, 3 Hz, pulse width ~250 ms, was used as the functional stimulus. A separate control scan was also performed where no stimulus was applied for either paradigm. Vibration frequency = 1kHz. In six of the seven animals, a localized increase in G' was observed in the somatosensory and motor cortex areas, whereas no difference was observed in the control scan. The average increase of G' = 14%. Two potential mechanisms were considered: (i) a vascular effect similar to BOLD in fMRI and (ii) calcium influx into the neurons. The first mechanism was ruled out based on results from an additional experiment where hypercapnia was induced to cause vasodilation. This implies the mechanism responsible is a primary measure of neuronal activation.

2:40

1pBAb4. The frequency-dependent effects of low-intensity ultrasound exposure on human colon carcinoma cells. Chloe Verducci, Hannah Seay, Janae Davis, Amy Longstreth, Yaa Kwakwa, and Maria-Teresa Herd (Phys., Mount Holyoke College, 50 College St., South Hadley, MA 01075, verdu22c@mtholyoke.edu)

Previous studies have established a correlative relationship between the acoustic properties of normal and malignant epithelial cells in response to high-frequency ultrasound exposure, indicating ultrasound's value as a tool in modern non-invasive cancer detection. More recently, ultrasound exposure has been extended into therapeutic fields, manipulated in frequency and power to stress and destroy specific cancer cells based on their determined acoustic properties. The present study seeks to determine the frequency-dependence of low intensity ultrasound exposure on varying cell types, beginning with colon carcinoma cells. The cells were exposed to a single element unfocused piezoelectric transducer at energies of less than 3 W/cm^2 and frequencies of 0, 5, 10, 15, 20, and 25 MHz, and evaluated for cytotoxicity indicated by lack of cell adhesion before and after treatment. Comparison is also made to non-cancerous human colon epithelial cells. This project compares the frequency-dependent effects of low intensity ultrasound on cells of varying cancerous and non- cancerous lineages, especially highlighting the low-point threshold.

3:00–3:20 Break

3:20

1pBAb5. Simulating fibrin clot mechanics using finite element methods. Brandon Chung Y. Yeung and E. Carr Everbach (Eng., Swarthmore College, 500 College Ave., Swarthmore, PA 19081, cyeung2@swarthmore.edu)

Blood clots inside blood vessels impede blood flow and can lead to blockage. Injection of thrombolytic agents affects the body systemically and may lead to hemorrhage. Risk is reduced with sonothrombolysis—high-amplitude pulsed ultrasound that drives micron-sized bubbles into violent oscillations, destroying the fibrin mesh of a clot. To gain insight into the 3D structure and mechanical behavior of fibrin clots, we fabricated a clot from purified fibrinogen, imaged it using a confocal microscope and 3D printed a plastic model of the image. Coordinates of fibrin connection points were entered into ANSYS, the clot finite-element-simulated for nodal displacements, and the bulk Young's modulus of the clot calculated. Simulations suggested that the elastic moduli in the three orthogonal directions were Ex = 113.5 Pa, Ey = 109.1 Pa, and Ez = 16.17 Pa. The close agreement between Ex and Ey supported the assumption of isotropy in a fibrin clot. The deviation of Ez from Ex and Ey could be attributed to the presence of the glass coverslip affecting the clot structure. Overall, results showed that confocal microscopy and simulations in ANSYS are useful in modeling clot structure and mechanics. Our next step is to simulate bubble activity inside the virtual clot.

3:40

1pBAb6. Numerical investigation of the subharmonic response of a cloud of interacting microbubbles. Hossein Haghi, Amin Jafari Sojahrood, Raffi Karshafian, and Michael C. Kolios (Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2k3, Canada, hossein.haghi@ryerson.ca)

Microbubbles (MBs) usually exist in polydisperse populations and often strongly interact with each other. Accurate investigation of the dynamics of the MBs requires considering the interaction between Microbubbles. We have developed an efficient method for numerically simulating N interacting MBs. The subharmonic (SH) responses of a polydispersions of 3-52 microbubbles with sizes between 2-5 microns, excited with ultrasound with a frequency and pressure of 1.8-8 MHz and 1-500 kPa were investigated. We show that if the frequency is set to the SH resonance of the larger MBs, the smaller MBs oscillations are controlled by the larger MB and the SH amplitude of the population of MBs increases. For small enough distances between MBs, one large MB may control the oscillations of the rest. If the excitation frequency is equal to the SH resonance of the smaller MBs, there exists two pressure regions. For lower pressures, the oscillations of the larger MB are out of phase with the smaller MBs and the resulting SH amplitude is smaller than the case without the bigger MB. As the pressure increases the oscillations of the larger MB becomes in phase with the smaller MBs and the SH response of the system is enhanced.

4:00

1pBAb7. Real-time monitoring and control of stable cavitation activity in pulsed sonication. Corentin Cornu, Matthieu Guédra, Jean-Christophe Béra, and Claude Inserra (Univ Lyon, Université Lyon 1, INSERM, LabTAU, F-69003, LYON, France, 151, cours Albert Thomas, Lyon 69424, France, corentin.cornu@inserm.fr)

Even if bubbles collapses are commonly thought to be the key element of cell permeabilization for drug delivery applications, recent works have demonstrated the possibility of transfecting cells by gentle oscillating bubbles (stable cavitation), possibly resulting in lower cell lysis or tissue damages. Nevertheless, in a bubble cloud, both stable and inertial cavitation activities would naturally coexist, thus making difficult to quantify the contribution of both regime on the drug delivery process. To distinguish each cavitation regime, a feedback-loop process is implemented on the subharmonic component emitted from the bubble cloud generated in a water tank by a focused transducer. This feedback loop, acting at a 250µs loop rate, allows adjusting the level of subharmonic emission as well as measuring inertial cavitation activity (broadband noise emission), by real-time modulating the applied voltage to the transducer. Evidences of control of the stable cavitation activity are reported, associated with (1) the possibility of exciting a time-stable subharmonic component, (2) the lowering of the broadband noise level and (3) the saving of acoustic energy (up to 20%) to ensure a given subharmonic emission level. [Work supported by the French National Research Agency, LabEx CeLyA (ANR-10-LABX-0060) and granted by the ANR-MOST project CARIBBBOU (ANR-15-CE19-0003).]

4:20

1pBAb8. Towards the accurate characterization of the shell parameters of microbubbles based on attenuation and sound speed measurements. Amin Jafari Sojahrood (Dept. of Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B2K3, Canada, amin.jafarisojahrood@ryerson.ca), Qian Li (Biomedical Eng., Boston Univ., Boston, MA), Hossein Haghi, Raffi Karshafian (Dept. of Phys., Ryerson Univ., Toronto, ON, Canada), Tyrone M. Porter (Biomedical Eng., Boston Univ., Boston, MA), and Michael C. Kolios (Dept. of Phys., Ryerson Univ., Toronto, ON, Canada)

Measurements of microbubble (MB) shell parameters is a challenging task because of the nonlinear dynamics of MBs. Shell parameter estimations that are typically based on solving linear models will generate inaccurate results, especially at higher pressure excitations. These approaches also often ignore the analysis of sound speed which provides useful information about the bulk modulus of the medium. In addition, the effect of MB-MB interaction is neglected. In this work, the attenuation and sound speed of monodisperse MB populations with mean diameters of 4 to 6 micron and peak concentrations of 1000 to 15000 bubbles/ml are measured for a pressure range of 10 to 100 kPa. The subharmonic pressure threshold of the solution was measured by narrowband excitations spanning from 1 to 4 MHz. The subharmonic generation pressure threshold was used to estimate an initial guess for shell viscosity and surface tension. The experimental results were fitted using numerical simulations of the Marmottant model and our recently developed nonlinear model for attenuation and sound speed. The effect of MB-MB interaction was also implemented using simulations of a lattice of interacting MBs (fitted to the measured sizes of MBs) to take into account the effect of concentration.

1p SUN. PM

Session 1pEA

Engineering Acoustics: Engineering Acoustics Topics I

Stephane Durand, Chair
LAUM, Université du Maine, LAUM - Université du Maine, Avenue Olivier Messiaen, Le Mans 72085, France

Contributed Papers

1:20

1pEA1. Reduced-size backing electrode microphone: Models and measurements. Cheng Qian, Alexey Podkovskiy (LAUM, Université du Maine, Le Mans, France), Petr Honzik (Faculty of Transportation Sci., CVUT, Praha 1, Czech Republic), Nicolas Joly, and Stephane Durand (LAUM, Université du Maine, LAUM - Université du Maine, Ave. Olivier Messiaen, Le Mans 72085, France, stephane.durand@univ-lemans.fr)

Reduced-size backing electrode microphones have been developed recently to achieve an easier match to specified response requirements. Such a development has used both analytical and numerical multi-physics modelings to validate the architecture efficiency. This microphone is composed of a membrane covering an annular cavity surrounding a central backing electrode. This simplified structure leads however to a higher sensitivity and an larger bandwidth, as shown in previous publications. A new modeling, based on lumped elements modeling, has been developed, in order to provide an easy a quick design tool for choosing the microphone parameters, these ones being fitted more precisely with a FEM modeling. These tools have leaded to the development of prototypes which have been characterized. The comparison of the measured data to the ones computed with the different models is presented. It shows a pretty good agreement between the several modelings and the experimental data, and it enlightens the need to take into account parasitic capacitance effects.

1:40

1pEA2. On acoustic characteristics of the Sound-7A coloring method. Yi Eun Young and Myungjin Bae (Information and TeleCommun., Soongsil Univ., Sangdo 1-dong, Dongjak-gu, Seoul, Seoul 156-743, South Korea, go6051@naver.com)

The most fundamental aspect of the noise influence on human beings is the size of the noise source. This paper investigates the acoustics characteristics of the sound-7A coloring system and compares each of the sound-7A colors with respect to a curve type. In this paper, the sound-7A coloring method is applied to the interlayer noise, when considering the level of noise and the energy ratio of the lightweight source to frequency band. The influence of the interlayer noise on cognitive mechanism of human beings is analyzed based on the loudness curve of human consciousness. The colors can be determined in both lightweight interlayer noise and heavy interlayer noise. As previously explained by the cocktail party effect human perceptive ability picks out some specific information among meaningless noise because it is somehow irritating to human ears. Out of some characteristics of the sound-7A coloring method, the energy ratio of the blue sound color band is nearly as twice as much for that of pink noise. This study can be utilized in our daily lives since it transfers aural perception into visual images such as illustrating lights in rainbow color coding and expressing musical instruments in special lighting performances.

2:00

1pEA3. On designing a new sound of the car-horn. SangHwi Jee, Myungsook Kim, and Myungjin Bae (Soongsil. Univ., Seoul 06978, South Korea, slayernights@ssu.ac.kr)

People are exposed to noise from birth to death. Hearing is considered the first human sense to awaken and the earliest to fall asleep. Human auditory sense distinguishes about 400,000 different sounds and classifies them. Noise is generally referred as unwanted sound, and human hearing is especially susceptible to noise. There are many kinds of noise such as household noise, construction site noise, transportation noise, noise between floors, and so on. Transportation noise, in particular, may harm more people since the driver as well as the people on the street are influenced by the noise. This study introduces a new friendly car-horn sound based on driver's preference through a MOS test over five different sounds. The sound has been selected based on characteristics of human auditory sense, the brainwave test, stress-index, and perceptive responses from one hundred subjects who participated in the experiment. The selected sound can be utilized in any motor vehicles available at the market so that not only the drivers honking but also the pedestrians hearing the car-horn sound can live more comfortably without hearing any annoying noise from the streets.

2:20

1pEA4. Spatially shaped acoustic transducer arrays. Stephen C. Butler, Thomas A. Frank, and Jackeline D. Diapis (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, stephen.c.butler1@navy.mil)

A shaped acoustic beam pattern that directs acoustic energy in the $\pm 45°$ direction with depressed acoustic energy at $0°$ is described. The shape of the beam pattern is controlled by area shading and the angular radius of the electro-acoustic transducer arrays. The advantage of such arrays is that acoustic energy is directed only in the required space needed at $\pm 45°$, thus minimizing the electrical input power requirements. As a result, the number of electrical wires and the driving circuit requirements is significantly reduced because the same electrical drive voltage can be applied to all the elements in the shaped array. The shaped array described herein can be developed in the form of a one-dimensional (1-D), two-dimensional (2-D), or three-dimensional (3-D) acoustic sonar transducer array (see US Patent 9,286,418 B1, Mar. 15, 2016). [Funding from ONR.]

2:40

1pEA5. Efficiency of capacitive micromachined ultrasound transducers For large signal non-collapsed operation. Amirabbas Pirouz (Elec. and Comput. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Love Bldg., Rm. 209, Atlanta, GA 30332, a.pirouz@gatech.edu) and F. Levent Degertekin (The George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Although capacitive micromachined ultrasonic transducers (CMUTs) are mostly considered for imaging applications for their broad bandwidth, these devices can also prove useful for high intensity focused ultrasound (HIFU) applications for therapeutics. For these purposes, energy conversion efficiency is especially significant for high intensity and high duty cycle

ultrasound applications. The usual small signal coupling coefficient analysis based on capacitance and resonance frequency measurement is not adequate for large signal application when nonlinearity comes to affect device performance. Therefore, an energy conversion ratio (ECR) based on a nonlinear large signal model has been proposed to analyze high power CMUT operation with and without DC bias (AC only). The results on a particular CMUT operating around 5 MHz show that AC only operation at half of device working frequency provides a higher level of pressure (0.8 dB more) as compared to DC biased case and can achieve 90% ECR. Since the input and output frequencies are not the same for AC only operation, Insertion loss (IL) is defined as the ratio of mechanical output power divided by the available electrical power in this case. With that definition, AC only and DC biased operation shows about the same IL in line with the ECR calculations.

3:00–3:20 Break

3:20

1pEA6. Comparison of different playback techniques in binaural head-related transfer function synthesis. Florian Wiese, Mina Fallahi, and Matthias Blau (Institut für Hörtechnik und Audiologie, Jade Hochschule Oldenburg, Ofener Str. 16-19, Oldenburg, Lower Saxony 26121, Germany, Florian.Wiese@jade-hs.de)

In binaural synthesis, the playback device (typically: headphones) can play an important role in achieving the desired perceptual authenticity. One potential source of error is the variability of headphone transfer functions when the headphones need to be taken off and on again after equalization. To avoid this issue, Erbes *et al.* (2012) proposed a device with miniature loudspeakers located about 5 cm away from the ears, which could remain in place. However, the device itself forms an obstacle not present in normal listening conditions and may therefore introduce direction-dependent artifacts. As an alternative, we propose a device which is placed in the subjects' ear canals. Due to its small size and the position in the ear canal, it is hoped that artifacts will be independent of the direction of sound incidence. In order to compare the two devices which remain in place to the more traditional playback over classical headphones, binaural syntheses were generated for 432 source positions (12 elevations and 36 azimuths) and related to transfer functions from real sources to microphones of a dummy head with ear canals. It was found that the agreement between synthesis and measurement was best with the in-ear device.

3:40

1pEA7. Smartphones as research platform for hearing improvement studies. Nasser Kehtarnavaz and Issa M. Panahi (Elec. Eng., Univ. of Texas at Dallas, 800 West Campbell Rd., Richardson, TX 75080, kehtar@utdallas.edu)

This poster presents the development of software tools at the University of Texas at Dallas to turn smartphones into a research platform for hearing improvement studies as part of the newly funded R01 NIH project entitled "Smartphone-Based Open Research Platform for Hearing Improvement Studies." The challenge in deploying smartphones as a research platform for hearing improvement studies lies in the use of programming languages that researchers are most familiar with. These languages are MATLAB and C. In other words, the challenge to deploy smartphones as a research platform is by not demanding researchers to know the programming languages associated with smartphones (Java for Android smartphones and Objective C for iOS smartphones) in order to turn smartphones into implementation research platforms. This challenge is met in this work by developing software shells that allow signal processing codes that are written in either C or MATLAB to be run on the ARM processors of smartphones/tablets. As part of this poster presentation, demos of the apps that have been generated so far based on these developed software shells will be presented.

4:00

1pEA8. Active control of a finite line source using multiple directional sources. Qi Hu and Shiu-Keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hung Hom, KLN, Hung Hom Na, Hong Kong, qi.bs.hu@connect.polyu.hk)

The active control of a finite line source in free space is studied using different types of secondary control sources to create a particular quiet zone. The simulations based on analytical formulation indicate that the active control improves significantly with the usage of directional secondary sources, such as axially oscillating baffled circular pistons. The comparison between directional sources of different directivity patterns shows that the directivity property has a determinant effect on the control results. A multi-part directional source, then, is introduced as the novel secondary control sources with its simplest two-part type. It consists of an inner-piston and an outer-concentric-annulus both oscillating axially with optimized amplitudes and relative phase. This novel secondary source helps to achieve excellent control results within a realistic physical size. The control performance regarding the novel sources with more outer annuluses are also studied.

4:20

1pEA9. Influence of piezoelectric materials on the performance of thin film hydrophones. Hanna Lewitz and Eckhard Quandt (Inst. for Mater. Sci., Kiel Univ., Kaiserstr. 2, Kiel 24143, Germany, hale@tf.uni-kiel.de)

Piezoelectric materials have been implemented in hydrophones in the early 20th century, mostly as cut bulk sensors or in cylindrical or spherical form made of piezoelectric materials [1]. Beginning with quartz soon other materials have been used, especially lead zirconate titanate with its high piezoelectric coefficients has been a large improvement in acoustic measurements. Nowadays miniaturization of the hydrophones is of interest, for example, in high resolution arrays for noise monitoring. Thus hydrophones materials are needed which have with less volume a similar performance as the state of the art bulk materials. Therefore, this work investigates hydrophones based on different piezoelectric thin films. These are mostly MEMS-production compatible and show good properties for the detection of acoustic signals. Therefore, sensors with different materials have been produced and the performance has been evaluated under laboratory conditions. [1] C. H. Sherman, J. L. Butler, Transducers and Arrays for Underwater Sound, Spring Sciences + Business Media, New York, 2007.

4:40

1pEA10. Electric power generation using acoustic helical wavefronts in air. Ruben D. Muelas H., Jhon F. Pazos-Ospina, and Joao L. Ealo (School of Mech. Eng., Universidad del Valle, Bldg. 351, Cali, Colombia, ruben.muelas@correounivalle.edu.co)

Acoustic Vortices (AV) are able to transfer angular momentum to matter and induce rotation to objects with different geometries. Several applications of this feature has already been reported. However, up to our knowledge, the possibility of generating electric power using this type of wavefronts has not been reported. In this work, we present experimental results on the electric power produced by a small generator coupled to a propeller of four blades that is insonified by an acoustic vortex beam of topological charge +1. The AV is produced using a multitransducer of 123 commercially available emitters driven with a continuous signal of 20 Vpp at 40 kHz in air. The propeller was located 40 mm far from the multitransducer, perpendicular to the principal axis of the AV. A voltage of 6 mV was observed at the electric terminals. A discussion on the obtained efficiency of the conversion is presented. Also, the possibility of harvesting residual acoustic energy by means of wave-matter exchange of linear and/or angular momentum is discussed. Special attention is paid to non-dissipative processes.

1p SUN. PM

5:20

1pEA11. Resonant gas oscillation in a tube of square cross section. Takeru Yano (Mech. Eng., Osaka Univ., 2-1, Yamada-oka, Suita 565-0871, Japan, yano@mech.eng.osaka-u.ac.jp)

Nonlinear resonant gas oscillations in a tube of square cross section are studied by solving the full system of Navier-Stokes equations for three-dimensional compressible gas flows with a finite-difference method. The nonlinear gas oscillations with and without shock waves and resulting time-averaged transport phenomena of mass, momentum and energy are investigated in the tube of square cross section. In particular, we find that there exist streamlines of acoustic streaming visiting both the inside and outside of boundary layer, which means that the mass, momentum and energy transports are confined neither in the inside nor in the outside of boundary layer on the tube wall, contrary to two-dimensional flows.

SUNDAY AFTERNOON, 25 JUNE 2017 ROOM 200, 1:15 P.M. TO 5:40 P.M.

Session 1pMU

Musical Acoustics and Architectural Acoustics: Concert Hall Acoustics

Jonas Braasch, Cochair
School of Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

David H. Griesinger, Cochair
Research, David Griesinger Acoustics, 221 Mt. Auburn St. #504, Cambridge, MA 02138

Chair's Introduction—1:15

Invited Papers

1:20

1pMU1. Influence of the exact source and receiver positions on the uncertainty of acoustical measurements in concert halls. Ingo B. Witew and Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany, ingo.witew@akustik.rwth-aachen.de)

Acoustical measurements are crucial to backing up theories or supporting conclusions in research and practical applications. In concert halls, however, it is well known that small changes to the receiver position yield a measurable change to the impulse response and the calculated single number parameter. This gives raise to the questions whether these spatial fluctuations limit the validity of measurements and whether there are implications for measurement applications? The presented study discusses how a measurement uncertainty approach may provide a new perspective to these problems. Based on array measurements a relationship has been established that quantifies how a change in measurement position leads to an average change to a room acoustic quantity. Strategies as outlined by the "Guide to the Expression of Uncertainty in Measurement" (GUM) are used to determine the bounds in which valid measurement results can be collected. It is discussed how these findings can be considered in applied measurement studies.

1:40

1pMU2. Disentangling room acoustics though binaural measurements: The importance of individual ear-canal variations. David H. Griesinger (Res., David Griesinger Acoust., 221 Mt. Auburn St. #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Evolution has endowed humans with extraordinary powers of hearing, powers that enable us to separate information-containing signals from noise and other signals with an acuity still unmatched by modern machines. We find that this ability depends critically on resonances in the ear canal. The ear canal and concha form a horn that boosts frequencies above 1000 Hz as much as 18 dB. These resonances are highly individual, headphones alter them, and just a 1 dB difference in the frequency balance at the eardrum can replace comprehension with confusion. We have developed a computer app that non-invasively equalizes headphones to accurately reproduce these resonances. Once headphones are individually equalized they reproduce binaural recordings with frontal localization and no head tracking. Using Lokki's anechoic recordings we can manipulate a single binaural measurement to create a binaural rendition of a small ensemble nearly identical to a live recording in the same seat. We can then lower or boost specific reflections and hear how perceptions of proximity, clarity, and envelopment change. We find that successful separation of the direct sound from reflections and reverberation is essential for both proximity and envelopment. The earliest reflections, whether medial or lateral, are almost always either inaudible or detrimental.

2:00

1pMU3. The use and abuse of early energy in concert space design. Christopher Blair and Paul H. Scarbrough (Akustiks, LLC, 93 North Main St., Norwalk, CT 06854, cblair@akustiks.com)

Fifty-five years ago Leo Beranek introduced the notion of a short initial time delay gap as being essential for acoustic "intimacy." However, in the authors' consulting experience, it has become apparent that many problems in concert and recital halls, onstage and in the audience, can be laid at the feet of too much early energy muddying clarity. A layman's description of the effect might be: "There's not enough 'air' around the sound." In support of this notion, recent research suggests that excessive early energy is actually the enemy of intimacy, masking the direct sound, modulating the phase of tonal components, inhibiting source localization. This paper presents the results of several listening experiments in halls of various sizes where adding absorption, venting or redirecting energy in specific critical locations dramatically enhanced the perception of both clarity and reverberation for musicians onstage and in the audience. Such methods often challenge traditional musician (and designer?) preconceptions, so whenever possible we employ quickly realized A/B comparisons to demonstrate treatment effectiveness.

2:20

1pMU4. Hybrid shaping applied to concert hall design. José A. Nepomuceno (Acústica & Sônica, Acustica & Sonica, Rua Fradique Coutinho, 955 cjt 12, São Paulo, São Paulo 05433-000, Brazil, info@acusticaesonica.com.br) and Christopher Blair (Akustiks, Norwalk, CT)

One of the first questions posed at the beginning of any concert hall design process is whether the room will be in the traditional long "shoebox" shape or the more intimate wrapping of the audience around the performers, characteristic of the "vineyard" approach. This paper presents the design process, acoustical measurements, and musician comments for a recently completed room where the answer was "both". Sala Minas is the home of the Orquestra Filarmônica de Minas Gerais, Belo Horizonte, Brazil. The concert hall has 1500 seats with a hybrid shape combining the attractive physical and acoustical attributes of shoebox and vineyard approaches. While the audience is held close to the performers, enhancing clarity and impact, the basic room geometry is a modified rectangle with a significant vertical "hard cap" zone providing ample reverberation, envelopment, and blending of orchestral sections. The result is a vineyard configuration with unusually consistent acoustic character in all the seating sections. Adjustable acoustic elements include a movable canopy over the stage, motorized acoustical banners, and shutters on the stage walls that can be opened or closed. After two seasons since Sala Minas opening, the reviews about its acoustics are impressive.

2:40

1pMU5. Applying subjective perception to acoustical planning. Gunter Engel (Müller-BBM, Robert-Koch-Str. 11, Planegg 82152, Germany, Gunter.Engel@mbbm.com)

Traditional room acoustical planning suffers from a considerable gap between the available measurement techniques and quality criteria at the one side and the subjective perception on the other side. Since the subjective perception is to a certain extent a matter of taste and moreover considerably overlaid by expectations and impressions which have nothing to do with acoustics, it is no wonder that it is so hard to find a suitable approach for creating the all over demanded perfect and world class acoustics. Applying new insights into the influence of early reflections on the perceived sound characteristics helps considerably to tailor the acoustics of a hall to your wishful thinking. The approach is illustrated by two concert halls with innovative design measures.

3:00

1pMU6. Achieving the excellent listening experience—Notes from 30 years experience in the successful integration of physical and electronic architecture. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

This paper will discuss important considerations in the elements that comprise electronic systems that alter perceived acoustics. In addition, the paper will discuss aspects of optimizing physical architecture to enable electronic architecture to work efficiently.

3:20–3:40 Break

3:40

1pMU7. Comparison of listener preference in concert halls from the stage and the audience. Samuel W. Clapp (Audio Information Processing, Tech. Univ. of Munich, Arcisstr. 21, München 80333, Germany, samuel.clapp@tum.de), Anne Guthrie (Arup Acoust., New York, NY), and Jonas Braasch (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

The perception of concert hall acoustics has been studied extensively from the perspective of the audience, but less extensively from the perspective of the musicians performing on stage. In this study, impulse response measurements were conducted in a group of eight concert and recital halls in the northeastern United States. A multi-channel microphone array was used to measure listening positions both on the stage and in the audience. The impulse responses recorded with the microphone array were used to generate auralizations for playback over multi-channel loudspeaker arrays, to investigate listeners' and musicians' preferences via listening tests. The audience positions were presented to the test subjects via static, pre-convolved auralizations. For the stage positions, all test subjects were musicians who, during the course of the test, performed on their instruments and heard the resulting auralizations from the stages of the different concert halls, generated in real-time. The results allowed for a comparison of listener preferences from the stage and audience positions in the same set of halls. The microphone array recordings also allowed for spatial energy analysis and the development of new spatial room impulse response parameters that could be correlated with listeners' preference judgments.

4:00

1pMU8. V.L. Jordan and Jørn Utzon: Acoustic and architectural interactions in the early design of the Major Hall at the Sydney Opera House, 1957-1962. Pamela Clements (Clements Acoust. Design Assoc., Unit 10, 23 Balfour Rd., Rose Bay, NSW 2029, Australia, clements.pamela@gmail.com)

Vilhelm Lassen Jordan was the first acoustician engaged by Jørn Utzon to work on the Sydney Opera House. Jordan was a Danish acoustic engineer whose advice was based in strictly scientific, quantifiable acoustics, combined with precedent. He was a pioneer in acoustic modeling. Jordan had a major influence on Utzon's first scheme for the Major Hall—a "rectangular " form showing Jordan's classic approach to acoustic design. Utzon's second, "faceted" scheme, developed between August 1960 and September 1963, was architecturally compelling but showed little of Jordan's influence. Jordan complained of the architect's "fancies" in the design, and by 1962 Utzon believed that Jordan had "given up." In June 1962, Utzon brought in Lothar Cremer and Werner Gabler from the Berlin Institute for Technical Acoustics, acousticians who provided scientific acoustic expertise and also worked with Utzon to integrate their acoustic recommendations into his design. This paper focuses on Jordan's early acoustic input into the design of the Major Hall, including his collaboration with Utzon on the first (rectangular) scheme and his early design of the second (faceted) scheme. It also considers dichotomies between science and art in the early design approaches of Jordan, Cremer and Gabler for the Major Hall.

4:20

1pMU9. Auditory perception in rooms. Jens Blauert (Inst. of Commun. Acoust., Ruhr-Univ. Bochum, Commun. Acoust., Bochum 44780, Germany, jens.blauert@rub.de) and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

In the design process of concerts halls, it is the task of the architects—preferably with the aid of experience acoustical consultants—to transform their concept on how the hall should sound into built form. To this end, a profound knowledge of the psychoacoustics of listening in concert halls is mandatory. Psychoacoustics relates auditory perception to the physical attribute of the sound field in the hall. While the sound field can be assessed by physical measurement, the measurement of auditory percept requires human assessors. However, to avoid costly listening test, algorithms been developed for estimating certain features of auditory percepts in concert halls. These estimators are useful for computer simulations of halls. In our talk, basic psychoacoustic phenomena and relevant instrumental estimators for psychoacoustic features will be evaluated. Finally, the concept of the "Quality of the Acoustics" of a concert hall will be considered in a broader context, thereby including aspects like the "Quality of Communication" in the hall. Type and quality of the information carrier in the light of the cognitive background of the listeners will be discussed, as well as the influence of visual, tactile, and olfactory cues on the quality assessment of concert halls.

4:40

1pMU10. Contemporary multi-use concert hall design: Experience and analysis. Anne Guthrie, Todd Brooks, Raj Patel, and Joe Solway (Arup, 77 Water St., New York, NY 10005, anne.guthrie@arup.com)

The acoustic design practice at Arup in New York has designed multiple performing arts spaces with a wide range of acoustic characteristics and functions, several of which have opened within the past few years. Many of these spaces make use of adjustable acoustics to accommodate a wide range of program within a single space, including acoustics control chambers, adjustable-height canopies, adjustable sound absorbing curtains and banners, and flexible configurations of various stage elements. Traditional objective parameters have been examined for both audience and stage acoustics, and additional spatial analysis is explored through visualization of 3D impulse responses. The relationships between these parameters and each venue's program goals will be addressed. Our experiences and findings obtained through the process of design, construction, and post-opening will be shared.

Contributed Papers

5:00

1pMU11. Methods to measure stage acoustic parameters: Overview and future research. Remy Wenmaekers, Constant Hak, Maarten Hornikx (Bldg. Phys. and Services, Eindhoven Univ. of Technol., P.O. Box 513, Eindhoven 5600MB, Netherlands, r.h.c.wenmaekers@tue.nl), and Armin Kohlrausch (Human Technol. Interaction, Eindhoven Univ. of Technol., Eindhoven, Netherlands)

The acoustics on stage has been recognized as an important design consideration for concert halls and other performance or rehearsal spaces. Stage acoustic parameters such as ST_{early} and ST_{late} are used to judge the early and late reflected sound levels on the stage. However, correlation of these parameters to perceptual attributes have not always been found. An explanation could be that the parameters used are not appropriate and/or that musicians find it hard to judge acoustic conditions. Another possible explanation is that the measurement methods used are not accurate enough. The goal of previous research by the authors was to investigate the uncertainties in the physical measurement. In this paper, an overview will be presented of the main findings that can serve as a starting point for future research. The following topics are covered: time windows for early and late sound, the reference level at 1 m distance, directivity of the omnidirectional sound source, impulse response quality, occupied stage measurements and directional transducers. Finally, a fast and musician friendly measurement method is presented that can be used to accurately measure acoustic parameters on a stage occupied by a full orchestra. The paper concludes with recommendations for future research.

5:20

1pMU12. Rich data trove gathered as a result of a unique opportunity with a capacity audience. David Greenberg (Creative Acoust., LLC, 5 Inwood Ln., Westport, CT, david@creative-acoustics.com), Steve Ellison, Melody Parker, and Roger W. Schwenke (Meyer Sound Labs, Inc., Berkeley, CA)

When an opportunity arises to perform an occupied room measurement, normally a short time is allotted to acquire data in order to minimize event disruption and audience discomfort. We therefore consider ourselves fortunate if one or two data sets are obtained under those conditions. An extremely rare—perhaps unique—opportunity arose from a confluence of factors: a 1600-seat concert hall at Liberty University, with a purposely over-sized area of movable curtains and banners to adjust the architectural reverberation; an installed active acoustics system, Constellation by Meyer Sound; a concert comprising distinct performances by orchestra, choir, and amplified worship band, with intermission to reset the architectural acoustics; and an interested client. The Constellation system provides 48 microphones distributed throughout the space, so a single set of sweeps through a system loudspeaker results in 48 measurements. The variables tested were

(a) adjustable acoustic absorption in and out; (b) full audience in and out; and (c) a nominal Constellation setting on and off. Measurement analysis both confirmed design intentions and elucidated the behavior of the two adjustable acoustic systems in the occupied and unoccupied hall. The architectural and active acoustic system designs, performance, measurements, and usage recommendations will be discussed.

SUNDAY AFTERNOON, 25 JUNE 2017 ROOM 203, 1:15 P.M. TO 4:00 P.M.

Session 1pNSa

Noise, Psychological and Physiological Acoustics, and Structural Acoustics and Vibration: Perception of Tonal Noise

Joonhee Lee, Cochair
Department of Building, Civil and Environmental Engineering, Concordia University, EV 6.231, 1515 Rue Sainte-Catherine O, Montreal, QC H3H1Y3, Canada

Roland Sottek, Cochair
HEAD acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

Chair's Introduction—1:15

Invited Papers

1:20

1pNSa1. Tonality calculation with modified DIN standard. Arne Oetjen and Steven van de Par (Acoust. Group, Carl von Ossietzky Univ. Oldenburg, Carl-von-Ossietzky-Straße 9-11, Oldenburg D-26129, Germany, arne.oetjen@uni-oldenburg.de)

Many natural sounds emitted by rotating machinery such as gearboxes, turbochargers, electrical motors or generators contain tonal parts. In the German standard "DIN 45691" a fft-based method for calculating the amount of perceived tonality in a complex sound is suggest that is based on the level difference between the tonal component and the noise in the surrounding critical band. Although the calculation method from DIN 45691 coincides well with subjective ratings for most stationary sounds, it often fails for sounds containing tonal components with rapid frequency changes such as turbochargers, due to the length of the analysis window. A pre-processing stage using shorter analysis windows was implemented that was still able to estimate precise frequencies and amplitudes of tonal components. The method is based on obtaining accurate frequency estimates of tonal components from dividing the spectra of the time differentiated signal and the original signal and is a modification of the method of [Desainte-Catherine and Marchand, J. Aud. Eng. Soc., 48, 654-667]. A post-processing stage using tracking and peak-picking algorithms was added to remove detected tonal components too short for being audible. This method allows both for high temporal and spectral resolution in the estimation of tonal components.

1:40

1pNSa2. Status quo of standardizing tonality calculation of stationary and time-varying sounds. Roland Sottek (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, roland.sottek@head-acoustics.de)

For many years in various applications of noise assessment, tonality measurement procedures such as the Tone-to-Noise Ratio (TNR), Prominence Ratio (PR), and DIN 45681 Tonality have been applied to identify and quantify prominent tonal components. Especially through the recent past as product sound pressure levels have become lower, disagreements between perception and measurement have increased across a wide range of product categories including automotive, Information Technology, and residential products. One factor is that tonality perceptions arising from spectrally elevated noise bands of various widths and slopes and from non-pure tones as well as from discrete (pure) tones, and from combinations of these, can be mis-measured or escape measure in "hybrid" sound pressure based tools and tools sensitive only to discrete tones. To address such issues, a new perceptually adequate tonality assessment method based on a hearing model of Sottek was developed which evaluates the nonlinear and time-dependent loudness of both tonal and broadband components, separating them via the autocorrelation function (ACF) and giving their spectral relationships. This new perception-model-based procedure, suitable for identifying and ranking tonalities from any sources, is proposed for the next edition of ECMA-74 as an alternative to the existing methods TNR and PR (ECMA-74, Annex D).

2:00

1pNSa3. Frequency tones and elevated decibel level of tones that are indicators of ill health or a state of health with associated pain. Bonnie Schnitta (SoundSense, LLC, 39 Industrial Rd., Unit 6, PO Box 1360, Wainscott, NY 11937, bonnie@soundsense.com) and Carter H. Sigmon (Dept. of Physical Medicine & Rehabilitation, Naval Medical Ctr. San Diego, San Diego, CA)

A correlation was confirmed in a study between noise sensitivity and ill-health. This study included clients that had hired an acoustic engineer to solve a noise problem, or patients who had a scheduled doctor's appointment. Male and female patients in diverse outpatient specialty clinics including: Sports Medicine, Pain Medicine, Hematology & Oncology and Breast Health were surveyed. The correlation shown of noise sensitivity to ill-health found was for a vast array of illnesses. A prior study was confirmed of a correlation of ill-health and noise sensitivity at lower frequencies that were typically only 1-2 dB(A) above ambient. Two additional findings were noted. First there was a heighted sensitivity to elevated noise levels of at least 15 dB(A) above ambient, such as construction noise, children screaming, sirens, etc. This noise sensitivity was found in people of various illnesses including but not limited to patients with cancer, atrial fibrillation, undescribed back pain, or untreated pain from a shoulder injury. The second finding was that some or all of the sensitivity dissipated upon treatment or remission of the illness or pain. This paper additionally discusses how to view this noise sensitivity as an indicator of ill-health, as well as influences on hospital design.

2:20

1pNSa4. Assessing tones in refrigeration and air-conditioning equipment. Derrick P. Knight (Ingersoll Rand - Trane, 3600 Pammel Creek Rd., La Crosse, WI 54601, derrick.knight@irco.com)

AHRI 1140 is a procedure for assessing the quality of sound for air-conditioning and refrigeration equipment. The procedure is based on work by Wells and Blazier in 1963 and jury listening tests by Penn State in 2001. This standard describes the measurements and calculations needed to determine the Sound Quality Index (SQI). The AHRI Technical Committee on Sound (TCoS) conducted informal surveying of manufacturing, design and consulting engineers which failed to identify any active users of SQI. TCoS has begun editing this standard in order to make it meaningful in identifying sound quality problems in a way that is useful to consultants and design engineers while being practical for manufacturers to adopt. The current effort is intended define a metric which will quantify the tonal characteristics of refrigeration and air-conditioning equipment. This presentation will briefly cover SQI and why TCoS believes SQI failed to achieve adoption. Additional tonal metrics will be reviewed, and potential difficulties with the existing metrics will be discussed. Most importantly, half of the presentation time will be used to allow audience feedback and suggestions.

2:40–3:00 Break

3:00

1pNSa5. The loudness of an amplitude-modulated sinusoid as a function of interaural modulator phase, modulation rate, and level. Brian C. Moore, Matthew Jervis, Luke Harries, and Josef Schlittenlacher (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

The aim was to test a model of loudness for binaurally presented time-varying sounds (Moore *et al.*, Trends in Hearing, in press). A 1000-Hz sinusoidal carrier was 100% sinusoidally amplitude modulated. The effect on its loudness of varying the interaural modulation phase difference (the IMPD) was assessed. The IMPD of the test sound was 90° or 180° and that of the comparison sound was 0°. A two-interval, two-alternative forced-choice method with a one-up/one-down rule was used to estimate the level difference between the test and the comparison sounds at the point of equal loudness (the LDEL) for baseline levels of 30 and 70 dB SPL and modulation rates of 1, 2, 4, 8, 16, and 32 Hz. The LDELs were negative (mean = -1.2 and -1.6 dB for IMPDs of 90° and 180°), indicating that non-zero IMPDs led to increased loudness. The model predicted that the LDELs should be most negative for modulation rates up to 4 Hz, and should be close to zero for the rate of 32 Hz. The data showed a pattern similar to the predicted pattern, but the LDELs for the modulation rate of 32 Hz were about 0.6 dB more negative than predicted.

3:20

1pNSa6. Can partial loudness of the tonal content be the basis for tone adjustments? Jesko L. Verhey and Jan Hots (Dept. of Experimental Audiol., Otto von Guericke Univ. Magdeburg, Leipziger Str. 44, Magdeburg 39120, Germany, jesko.verhey@med.ovgu.de)

Environmental sounds containing tonal components are more annoying than those without audible tonal components. This is considered in several standards addressing the assessment of noise immissions. These standards have in common that the strength of the tonal component, referred to as tonality, tonalness, or magnitude of tonal content is taken as the level of the prominent audible tone relative to the surrounding background noise. Based on this magnitude of tonal content, some standards add tone adjustments to the measured sound levels to account for the reduced acceptance of a sound with audible tonal components. On the basis of experimental data and model predictions, the present study shows that the magnitude of the tonal content is better characterized by its partial loudness than by the signal-to-noise ratio of the prominent tonal component. Partial loudness of the tonal component may be considered in future standards as a basis for the assessment of the annoyance of the tonal content of a sound and thus the determination of a tone adjustment.

3:40

1pNSa7. Uncertainty in tone quantification methods of background noise for enclosed spaces. Joonhee Lee (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., EV 6.231, 1515 Rue Sainte-Catherine, Montreal, QC H3H1Y3, Canada, joonhee.lee@concordia.ca) and Lily M. Wang (Durham School of Architectural Eng., Univ. of Nebraska - Lincoln, Omaha, NE)

The noticeable tones in background noises can annoy and disturb human listeners. The noise community has been developing methods to quantify perception of tones and lately trying to propose new tone guidelines to regulate the maximum level of tones in noise. Prior to proposing guidelines, the reliability of existing tone quantification methods should be examined. Thus, this paper investigates uncertainty of the tone quantifying methods from ANSI or ISO standards including Tonal Audibility, Prominence Ratio, and Tone-to-Noise Ratio. This study will discuss major causes of uncertainty in measuring the tones from using the methods. It will cover definitions of tones in each method, how these metrics separate tones and broadband noises, and how they analyze signals in time and frequency domains. Lastly, this paper will also investigate effects of room modes on the measured tonality in indoor environments. The variances of the measured tonality across measurement positions for the assorted metrics will be presented. Acceptability of the variances in tonality will also be discussed.

SUNDAY AFTERNOON, 25 JUNE 2017 ROOM 202, 1:15 P.M. TO 5:40 P.M.

Session 1pNSb

Noise and Physical Acoustics: Session in Honor of Kenneth Plotkin

Ben H. Sharp, Cochair
Ben Sharp Acoustics, 7802 Trammell Rd., Annandale, VA 22003

Juliet Page, Cochair
Environmental Measurement and Modeling, Volpe National Transportation Systems Center, 55 Broadway, Cambridge, MA 02142

Victor Sparrow, Cochair
Grad. Program in Acoustics, Penn State, 201 Applied Science Bldg., University Park, PA 16802

Philippe Blanc-Benon, Cochair
Centre acoustique, LMFA UMR CNRS 5509, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, Ecully 69134 Ecully Cedex, France

Chair's Introduction—1:15

Invited Paper

1:20

1pNSb1. Kenneth J. Plotkin—A most unforgettable character. Ben H. Sharp (Ben Sharp Acoust., LLC, 7802 Trammell Rd., Annandale, VA 22003, bhs940@yahoo.com)

In a career lasting over 45 years, Ken Plotkin established himself as a world leader in aeroacoustics, best known for his research studies of sonic boom. He developed the first practical method of predicting focused sonic booms and has been a key participant in recent studies to mitigate sonic booms, for which he has been awarded several NASA awards for excellence. But Ken's contributions to the field of acoustics have been much more diverse than many realize. They have included topics such as highway, vehicle and tire noise, psychoacoustics, aircraft noise simulation modeling, soundscape analysis and monitoring, and community noise. Ken was one of the most intelligent people the author has ever met, and one of the most inventive. He had the ability to break down complex problems into simple components that could be understood and solved, often using limited available data and simple theoretical models. Refreshingly modest with a self-deprecating and cynical sense of humor, Ken was one of those rare people who, once met, would never be forgotten. As a long-term colleague and friend, the author will discuss some of his major contributions and share some of his experiences working with him.

Contributed Papers

1:40

1pNSb2. Memorable interactions with Dr. Ken Plotkin during NASA sonic boom research. Peter Coen (NASA, NASA Langley Res. Ctr., MS 264, Hampton, VA 23681, peter.g.coen@nasa.gov)

Dr. Ken Plotkin was an active and vital contributor to numerous NASA research activities over the course of his career. His involvement clearly advanced NASA's objective and improved the state of knowledge of the prediction and reduction of sonic boom noise, but it also created many fond and amusing memories for those fortunate to work with him. This remembrance of Ken will recall some of his intellectual and comical contributions.

2:00

1pNSb3. Dr. Kenneth J. Plotkin: Friend and mentor. Joseph A. Salamone (P.O. Box 1372, Tybee Island, GA 31428, joesalamone3@gmail.com)

Dr. Kenneth J. Plotkin was the chief scientist at Wyle and was known in the acoustics community as an expert in many aspects of aircraft and transportation noise. He was also widely regarded as an authority on the subject of sonic booms. His enthusiasm for research and new discoveries was contagious—it was almost impossible not to be influenced by his example. He was very willing to share his expertise with those who also ventured into the realm of sonic boom propagation. It is an honor to be one of the people who benefited from his willingness to share his wisdom and learn from his teaching. An initial discussion with Dr. Plotkin inquiring about his knowledge of the rise time of shocks in sonic booms developed into friendship and mentoring that lasted almost 14 years. This presentation will highlight these early conversations and share how they evolved over time. Additionally, the impact he made on my engineering career and graduate school education will also be presented. [Acknowledgments to the National Aeronautics and Space Administration, Federal Aviation Administration, The Pennsylvania State University, Wyle, and Gulfstream Aerospace Corporation.]

2:20

1pNSb4. Dr. Kenneth Plotkin's myriad contributions to the National Aeronautics and Space Administration's supersonic mission. Edward A. Haering (Res. AeroDynam., NASA Armstrong, M.S. 2228, PO Box 273, Edwards, CA 93523, edward.a.haering@nasa.gov)

The world as a whole, and NASA, in particular, owes a large debt of gratitude to Dr. Kenneth Plotkin for his decades of service in the field of sonic boom research and advancement of quiet supersonic transportation. This presentation will highlight the contributions of Dr. Plotkin to a myriad of NASA projects. One of the largest efforts was the assembly and continual improvement of sonic boom propagation software tools, collectively called PCBoom, which allowed the analysis of real and imagined vehicles from Mach cutoff conditions to the hypersonic. He was a driving force behind reshaping aircraft to demonstrate quieter sonic booms, first with the plans for a modified Firebee drone and SR-71, and then with the highly successful Shaped Sonic Boom Demonstrator series of flights. Dr. Plotkin's partnership with NASA Armstrong resulted in the development of the low boom dive maneuver to allow quiet sonic boom testing on structures and people using existing aircraft, as well as a sonic boom cockpit display that has recently been tested in flight. Dr. Plotkin was also instrumental in such research campaigns as SCAMP, WSPR, and FaINT. Throughout all, Dr. Plotkin's phenomenal intellect, tireless dedication, and irreverent humor made working with him a joy.

2:40

1pNSb5. Recollections from four decades of sonic boom research with Dr. Kenneth J. Plotkin. Domenic J. Maglieri (Eagle Aeronautics, Inc., 732 Thimble Shoals Blvd.'Bldg. C 204, Newport News, VA 23606, sonicboomexpert1@verizon.net)

Every so often the technical community is blessed to have a very special & talented individual join its folds. In the past half century, the issue of sonic boom needed such a talent who would provide leadership and pioneering contributions to the understanding of the sonic boom, its generation, propagation, prediction, and minimization and its effect on people and structures. Ken was that person. This presentation is of a personal note touching on a few recollections of working with Ken for over 45 years. It will begin with my introduction to Ken in 1970, highlight a key ingredient to his 1971 doctorate thesis & reflect on the significance of his findings, his initial reaction to the 1990 review of his AIAA Journal of Aircraft paper, the coauthoring of a chapter on sonic boom in 1991,our views on need to demonstrate the persistence of a shaped signature, & whether signature shaping will minimizing the transition focus boom, our discussions on whether booms are observed from subsonic flight, & a listing of the boom efforts we enjoyed & publications we coauthored. I miss Ken, his enthusiasm & entertaining ways, his brilliance. I believe he deserves a place alongside G. B.Whitham, A. R.Seebass, & A. R.George.

3:00–3:20 Break

Invited Papers

3:20

1pNSb6. Kenneth Plotkin——Military noise and sonic boom. Micah Downing (Blue Ridge Res. and Consulting, 29 N. Market St., Ste. 700, Asheville, NC 28801, micah.downing@blueridgeresearch.com)

As a young researcher at the Air Force Research Laboratory, I was blessed with the good fortune of working with Ken on a variety of military sonic boom and aircraft noise projects. His mentorship on sonic boom theory required patience on his part but resulted in a successful collaboration, which included an improved PCBoom model and focused sonic boom measurements. From this work, we demonstrated how to make sonic booms louder unlike his later efforts to help minimize future aircraft's sonic booms. Ken also led major upgrades to the military's aircraft noise model, NoiseMap. His work resulted in the development of the simulation noise model, NMSim and its noise animations. Through Ken's efforts, we now have tools to better explain sonic boom and aircraft noise. As my mentor, I hope to share some highlights of working with Ken.

3:40

1pNSb7. Personal memories of Ken Plotkin. Nicholas P. Miller (Harris Miller Miller & Hanson Inc., 77 S. Bedford St., Burlington, MA 01803, nmiller@hmmh.com)

Ken and I were competitors, co-participants in conferences and professional societies and, occasionally, co-workers. Our times together at meetings or on projects covered about ten years starting in the late 1990s. We worked together on projects for the National Park Service, and spent many meetings of SAE Committee A21 together. These times permitted us to become well-acquainted with each other, and to develop mutual respect for each other's experience and knowledge. We were almost exact contemporaries, being the same ages and starting our careers at almost the same time—Ken at Wyle and I at BBN. Our experiences were somewhat different, with Ken working mostly on military noise issues and I on general transportation noise. We each worked with expert mentors on the our respective coasts; Ken working with Lou Southerland and others in California, I with Ted Schultz, Dick Bolt, and others in Massachusetts. From these times, we enjoyed each other's company, learned a lot from each other, and made memories.

4:00

1pNSb8. Kenneth J. Plotkin's contributions to acoustics and sonic boom research at NASA. Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov)

Dr. Kenneth J. Plotkin played a significant role in furthering knowledge related to outdoor sound propagation and, in particular, propagation of sonic booms from supersonic aircraft. This presentation focuses on Dr. Plotkin's support of NASA's research in these areas. In recent years, he worked with NASA on expanding the capabilities of the modeling software PCBOOM for prediction of sonic boom propagation for a variety of aircraft flight conditions in complex atmospheric environments. Dr. Plotkin was also instrumental in the planning, execution, and analysis of several NASA supersonic flight test campaigns aimed at gathering data for development and validation of prediction models like PCBOOM. NASA's sonic boom research has benefited greatly from these collaborations, and it is our hope to honor his legacy by continuing work in this area.

4:20

1pNSb9. Overview of acoustic and sonic boom advancements during development of NASA launch vehicles. Janice Houston (Marshall Space Flight Ctr., NASA Marshall Space Flight Ctr., Huntsville, AL 35812, janice.d.houston@nasa.gov), Jess Jones (AI Signal Res. Inc., Huntsville, AL), R. Jeremy Kenny, Tomas Nesman, Darren Reed (Marshall Space Flight Ctr., Huntsville, AL), and Bruce Vu (Kennedy Space Ctr., Melbourne, FL)

During the study and development of NASA space vehicles, acoustic environments have been a critical design input. This paper surveys some of the key challenges and focuses on the contributions and collaborations of Kenneth J. Plotkin/Wyle Laboratories with various NASA centers and personnel. In the mid-1960's and early 1970's, a method for predicting in-flight fluctuating environments for vehicle systems was developed for the Saturn Development Programs at NASA Marshall Space Flight Center (MSFC). With the Space Shuttle Vehicle development in the 1970's, sonic boom became a concern and sonic boom focusing was studied. Attention was turned to the Space Shuttle Orbiter entry maneuvers during the approach to the KSC landing site. In 1993, a PC version for sonic boom prediction was developed for the National Launch System study. Near-field pressure data from computational fluid dynamics analyses were used to develop the shape factors used in the X-33 sonic boom analyses. For the X-34 sonic boom analyses, the influence of plumes on the shape factor was included. In the 2000s, rocket noise prediction software at the KSC launch platform was developed for the Constellation Program. All this acoustic work is being leveraged on NASA's latest vehicle, the Space Launch System.

4:40

1pNSb10. Remembering Ken Plotkin: Colleague, mentor, and friend. Juliet Page (Environ. Measurement and Modeling, Volpe National Transportation Systems Ctr., 55 Broadway, Cambridge, MA 02142, juliet.page@dot.gov)

Dr. Kenneth J. Plotkin's contributions to the field of acoustics are numerous and cover a variety of areas such as sonic boom, jet noise, community noise, and atmospheric propagation. I will review many of Ken's contributions to the field of acoustics, including shared experiences during unique and challenging projects and field measurement programs. I will also highlight Ken's leading role in the development of acoustic visualization techniques.

5:00–5:40 Panel Discussion

Session 1pPA

Physical Acoustics and Biomedical Acoustics: Acoustofluidics II

Jürg Dual, Cochair
ETH Zurich, Tannenstr. 3, Zurich 8092, Switzerland

Charles Thompson, Cochair
ECE, UMASS, 1 Univ. Ave., Lowell, MA 01854

Max Denis, Cochair
U.S. Army Research Lab., 2800 Powder Mill Road, Adelphi, MD 20783-1197

Invited Papers

1:20

1pPA1. Macro-scale cell manipulation using bulk ultrasonic standing waves for biopharmacy and cellular therapy applications. Bart Lipkens, Kedar C. Chitale, Benjamin P. Ross-Johnsrud, and Walter Presz (FloDesign Sonics, 1215 Wilbraham Rd., Box S-5024, Springfield, MA 01119, blipkens@wne.edu)

Acoustic standing wave fields are widely used in MEMS applications to manipulate micron sized particles in fluids with typical fluid channel dimensions of half a wavelength. This report presents three novel acoustofluidic platforms for particle separation and/or manipulation at macroscale, i.e., tens to hundreds of wavelengths. The first platform uses multidimensional standing waves which generate lateral radiation forces that trap and tightly cluster suspended fluid or particulate, enhancing the gravitational settling effect that results in continuous, macroscale separation. The second platform employs acoustic radiation forces generated near the edge of an acoustic standing wave to hold back particles and generate a wall type separation effect. The third platform uses the acoustic radiation forces generated by a macroscale, angled standing wave to deflect particles in a controlled fashion for particle manipulation and/or differentiation. Applications are focused in biopharmacy and cellular and gene therapy: mammalian cell clarification, continuous perfusion of bioreactors, cell concentration and washing, cell sorting and differentiation, fractionation, microcarrier-cell separation, and affinity acoustic separation. A commercial cell clarification device has been introduced. The key physics principles related to acoustic radiation force and low Reynolds number multi-phase flows are discussed. Experimental results of cell clarification, perfusion, and manipulation are shown.

1:40

1pPA2. Acoustofluidic manipulation of biological bodies: Generation, visualization, and stimulation of cellular constructs. Dario Carugo, Bjorn Hammarström, Umesh Jonnalagadda, Junjun Lei, Filip Plazonic, Walid Messaoudi, Zaid Ibrahim Shaglwf, Peter Glynne-Jones, and Martyn Hill (Eng. Sci., Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, d.carugo@soton.ac.uk)

Ultrasound-based external manipulation of biological bodies in microfluidics has emerged as a contactless way of manipulating cells and particles for a range of applications, including sample enrichment, filtration, and sorting. Furthermore, it has been recently utilized to drive cells to form multi-cellular architectures, including clusters and planar sheets, by appropriately designing the resonant ultrasound field within the acoustofluidic device. In this presentation, we demonstrate the development of ultrasonic bioreactors for generating 3D, scaffold-free tissue constructs. We apply this technology to the generation of neocartilage grafts and examine their potential for repair chondral defects, and to the generation of co-culture models of the mucosal airway. Furthermore, we illustrate how the ultrasonic standing wave field can be designed to generate and modulate different stress regimes on suspended cells, for activating mechanotransductive pathways or for enhancing intracellular delivery of bioactive compounds. Integration of acoustofluidic systems with advanced microscopy techniques for quantifying biophysical effects of ultrasound on single cells or cellular constructs is also discussed.

2:00

1pPA3. Acoustofluidic manipulation of biological bodies: Applications in medical and environmental diagnosis. Dario Carugo, Bjorn Hammarström, Umesh Jonnalagadda, Junjun Lei, Filip Plazonic, Walid Messaoudi, Zaid Ibrahim Shaglwf, Peter Glynne-Jones, and Martyn Hill (Eng. Sci., Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, J.Lei@soton.ac.uk)

Ultrasound-based external forcing of biological bodies in microfluidics has emerged as a contactless way of manipulating cells and particles for a range of applications, including sample enrichment, filtration, and sorting. Recently, acoustic radiation forces have shown potential for manipulating pathogenic organisms in biological assays. In this presentation, we demonstrate the development of acoustofluidic systems designed for high-throughput manipulation and capturing of biological bodies in applications ranging from medical to environmental diagnosis. Specifically, we apply our acoustofluidic systems to the detection of (i) cancer and immune cells in the early-stage diagnosis of blood malignancies and allergies, and (ii) bacterial microorganisms, spores, and planktonic cells for screening of environmental and industrial samples.

2:20

1pPA4. Macroscale angled ultrasonic standing waves: A novel approach for particle manipulation. Kedar C. Chitale, Walter Presz (Flodesign Sonics, 380 Main St., Wilbraham, MA 01095, k.chitale@fdsonics.com), Bart Lipkens (Flodesign Sonics, Springfield, MA), Benjamin P. Ross-Johnsrud, Miles Hyman, and Marc Lamontagne (Flodesign Sonics, Wilbraham, MA)

Macro scale acoustophoretic devices use radiation forces to trap particles inside a standing wave to separate them from a mixture in a continuous fashion. However, these devices are limited by factors such as flow rates, residence times, and temperature rise which could be detrimental for certain applications. A novel method of separating, sorting and differentiating various particles using bulk angled ultrasonic standing waves is presented. This technique offers very sensitive separation capability with respect to size and acoustic contrast of particles. Universal curves are developed for particle deflection from the bulk flow direction at all wave angles as a function of a non-dimensional parameter defined by the ratio of acoustic radiation force to viscous drag force. Both CFD (Computational Fluid Dynamics) and model test data verify the analytical predictions. New macroscale, ultrasonic separator concepts are presented that use the angle wave technology to effectively deflect and/or separate microcarrier beads from a flowing mixture at high speeds when compared to conventional ultrasonic separators. Model test data verify the ability to move, differentiate, separate, or fractionate particles in suspension by size and acoustic contrast.

2:40

1pPA5. Acoustic edge effect: Novel acoustophoretic cell retention to enable continuous bioprocessing. Benjamin P. Ross-Johnsrud, Erik Miller, Hayley Hicks, Kedar C. Chitale, Walter Presz (FloDesign Sonics, 380 Main St., Wilbraham, MA 01095, b.johnsrud@fdsonics.com), and Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA)

There is currently a shift in Bioprocessing towards continuous manufacturing of monoclonal antibodies or recombinant proteins in perfusion mammalian cell cultures (Konstantinov & Cooney, Journal of Pharmaceutical Sciences, 2015). A cell retention device is the key technology component that enables the shift to continuous production. A novel acoustic cell retention device operates by continuously drawing off a harvest flow, equal to the perfusion rate of the bioreactor, while recirculating the retained cells back to the bioreactor. The harvest flow path is tangent and significantly smaller than the recirculation rate. The device utilizes a novel acoustophoretic effect known as an "acoustic edge/interface" effect in conjunction with a recirculating flow beneath the acoustic harvest chamber which collects and returns cells to the bioreactor. This interface effect operates by creating a radiation pressure/force field at the interface between cell-free harvest and cell-laden circulating fluids. Numerical results show an insight into the mechanism of the acoustic edge effect. Experimental results confirm the existence of this novel acoustic edge effect. CHO cell perfusion cultures were operated continuously for $>$15 days. This technology delivers continuous cell retention and steady unhindered product transmission enabling continuous production of biopharmaceuticals unlike traditional hollow-fiber tangential flow filtration.

3:00

1pPA6. Acoustophoresis mediated chromatography processing: Capture of proteins from cell cultures. Thomas Kennedy, Malcolm Pluskal, Rudolf Gilmanshin (FloDesign Sonics, 380 Main St., Wilbraham, MA 01095, t.kennedy@fdsonics.com), and Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA)

Chromatographic purification of target biomolecules is an important downstream process step in the development of new therapeutic agents, such as antibodies. This step employs chromatographic processes, such as affinity separation utilizing Protein A, ion exchange, or mixed mode chemistries. The workflow typically involves a packed column(s) and several chromatographic steps to achieve the desired level of purity. The steps can be time-consuming and expensive. A new process will be described, employing an acoustic standing wave in a fluid chamber to partition and maintain solid phase beads in an acoustically fluidized bed format to capture, wash, and elute the target biomolecule. Purification workflow(s) will be described for the following applications: (1) capture of a monoclonal antibody by Protein A beads from a crude cell culture system, (2) capture of recombinant Green Fluorescent protein (GFP) by anion exchange from a crude cell lysate. The workflow will include wash step(s) and recovery with a specific elution conditions. This communication will clearly demonstrate that an acoustophoresis process can purify proteins without a packed chromatography column. This new approach will minimize the time and cost involved in current purification workflows.

3:20–3:40 Break

3:40

1pPA7. Boundary interactions with vortical disturbance in an acoustofluidic channel. Kavitha Chandra, Charles Thompson, and Vineet Mehta (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, kavitha_chandra@uml.edu)

In this work, mechanisms governing the generation of unstable vortical disturbances and their spatiotemporal characteristics are examined. Time-harmonic boundary and pressure driven flows are of particular interest. In the inner region near the solid-fluid interface, the vortical components of the particle velocity are taken to behave incompressibly. To accommodate time-dependent channel wall geometries a curvilinear coordinate based pseudo-spectral method is developed. The method allows for the direct numerical solution of the three-dimensional time-dependent Navier-Stokes equation for high streaming Reynolds conditions. The conditions for centrifugal destabilization and transition are examined.

4:00

1pPA8. Ultrasonic robotics in microfluidic cavities. Jürg Dual, Michael Gerlt, Philipp Hahn, Stefan Lakaemper, Ivo Leibacher (ETH Zurich, Tannenstr. 3, Zurich 8092, Switzerland, dual@imes.mavt.ethz.ch), Andreas Lamprecht, Peter Reichert, Nadia S. Vertti Quintero, Xavier Casadevall i Solvas, Rudiyanto Gunawan , and Andrew deMello (ETH Zurich, Zurich, Zurich, Switzerland)

Ultrasonic standing waves are often used in biomedical applications. It has become quite common to move beads, cells, droplets, and other particles for sorting or biomedical analysis in microfluidic cavities by bulk acoustic waves or by vibrations excited by piezoelectric transducers. The motion of particles is determined by streaming and radiation forces. For the calculation of the radiation forces acting on single particles Gorkov's potential is considered to be the modeling tool of choice, once the acoustic field and the properties of constituents (fluid and particle density and compressibility, respectively) are known. For the acoustic streaming, predictions can be made numerically. For both aspects large uncertainties exist, due to the complexity of the system and fluid structure interaction at multiple levels. In this paper, first various characterization tools for the acoustic field in the cavity are described. They consist of the interplay between numerical modeling of the device, impedance analysis of the piezoelectric transducer used, interferometric analysis of surface displacements and an optical trap to measure the forces on the particles directly. Secondly, a number of recent applications are shown, including the sorting and immobilization of *C. elegans.* Furthermore, fascinating behavior of multiple interacting particles is reported.

4:20

1pPA9. Acoustic nonlinearity and the generation of large tensile pressures to explain atomization in drop-chain acoustic fountains. Oleg Sapozhnikov (Phys. Faculty, Moscow State Univ., and CIMU, Appl. Phys. Lab., Univ. of Washington, Leninskie Gory, Moscow 119991, Russian Federation, oleg@acs366.phys.msu.ru), Elena Annenkova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Wayne Kreider (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

An ultrasound beam propagating upward in a liquid creates an acoustic fountain at a gas interface in the form of a drop chain. High-speed photography shows that one or several drops in such a fountain explode in less than a millisecond, resulting in liquid atomization [Simon *et al*. J. Fluid Mech., 2015, 766, pp. 129-146]. To explain this phenomenon, a nonlinear theory involving an isolated spherical drop is developed. The model considers an initial excitation in the form of a spherical standing acoustic wave at the lowest resonance frequency, i.e., when the drop diameter coincides with a wavelength. If higher harmonics are generated inside the drop due to acoustic nonlinearity, these harmonics will also have the form of standing spherical waves. At higher frequencies, more of the energy of each harmonic is localized near the drop center. Calculations demonstrate that harmonic generation can lead to large increases in both peak positive and peak negative acoustic pressure at the drop center. Such large tensile pressures may exceed the intrinsic cavitation threshold, leading to the nucleation of a bubble at the center and explosion of the drop as the bubble grows rapidly. [Work supported by RFBR 17-02-00261 and NIH R01EB007643.]

4:40

1pPA10. Acoustic characterization of microbubble clouds by attenuation and celerity spectroscopy. Lilian D'Hondt (Nuclear Technol. Dept., French Atomic Energy Commission, CEA Cadarache, DEN/CAD/DTN/STCP/LIET - Bât. 202, Saint Paul Lez Durance 13108, France, lilian.d'hondt@cea.fr), Cedric Payan, Serge Mensah (Aix Marseille Univ, CNRS, Centrale Marseille, LMA, Marseille, France), and Matthieu Cavaro (Nuclear Technol. Dept., French Atomic Energy Commission, Saint Paul lez Durance, France)

In 4th generation nuclear reactors cooled with liquid sodium, argon microbubbles are present in the primary sodium. Due to the opacity of liquid sodium, acoustic control methods are chosen for operating inspections but this bubble presence greatly affects the acoustical properties of the medium. It is therefore required to characterize the microbubble cloud, i.e., to provide the bubble's volume fraction and the size distribution. Safety demands the proposed method to be robust and applicable with few assumptions (about the bubble populations) as possible. The objective of this study is to evaluate the performance of spectroscopic methods (based on celerity and attenuation) in the presence of bubbles whose sizes and surface (or volume) contributions are very different. Two methods of evaluating the histogram and the void fraction are compared. The first is based on the inversion of the integral equation of the complex wave number derived by Commander and Prosperetti. The second, which assumes the populations to have log-normal or sums of Gaussians distributions, performs an adjustment of the distribution's parameters to fit attenuation and celerity curves measurements. These methods are compared with experimental data obtained using ACWABUL facilities at CEA Cadarache.

5:00

1pPA11. Experimental analysis of backscattering by cylindrical shell with internal plate at oblique incidence. Yunzhe Tong, Bin Wang, and Jun Fan (Shanghai Jiao Tong Univ., 800 Dongchuan Rd., Minhang District, Shanghai 200240, China, tongyunzhe@sjtu.edu.cn)

Through an experimental approach, this paper studies the flexural wave coupling on a cylindrical shell with internal structural. Impulse response backscattering measurements are presented and interpreted for the scattering of obliquely incident plane waves by a fluid-loaded stiffened cylindrical shell and a corresponding empty cylindrical shell respectively. The stiffened cylindrical shell is reinforced by a thin internal plate which is diametrically attached to the shell along its axial direction. The time series data are fast Fourier transformed and the modulus normalized according to the direct wave spectrum. Result are plotted as frequency-angle spectra. Compared with that from the corresponding empty cylindrical shell, the subsonic flexural waves on the cylindrical shell will interact between the attachments and some of their energy can be converted into radiating waves.

5:20

1pPA12. T-matrix method implementation for acoustic Bessel beam scattering from elastic solids and shells. Zhixiong Gong (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Webster Physical Sci. 754, Pullman, WA 99164-2814, zhixiong.gong@wsu.edu), Philip L. Marston (Dept. of Phys. and Astronomy, Washington State Univ., Pullman, WA), Yingbin Chai, and Wei Li (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei, China)

T-matrix method (TMM) has been demonstrated to be an effective tool for the application of acoustic Bessel beam (ABB) scattering from rigid shapes, owing to the fact that the incident ABBs could be appropriate to expand on the basis of spherical harmonics [Gong *et al*., J. Sound Vibr. 383, 233-247 (2016)]. In this work, we try to extend the TMM to further calculate ABB scattering from complicated elastic shapes, spheroids/ spheroidal shells for instance. Some numerical techniques are successfully implemented to overcome the instability problem during matrix inversion procedures for nonspherical shapes. Resonance scattering theory and ray theory [Kargl and Marston, J. Acoust. Soc. Am. 88, 1103-1113 (1990)] are employed to explore and interpret several novel properties of scattering from elastic shapes illuminated by ABBs, thus revealing the corresponding mechanism of scattering by ABBs. Furthermore, the present work will perform as a foundation work to extend the applicability of TMM to study acoustic radiation force and torque in ABBs in future. [Work supported by NSFC.]

Session 1pPPa

Psychological and Physiological Acoustics and Speech Communication: Honoring the Contributions of Louis Braida to the Study of Auditory and Speech Perception

Charlotte M. Reed, Cochair
Research Laboratory of Electronics, Massachusetts Institute of Technology, Room 36-751, MIT, 77 Massachusetts Ave., Cambridge, MA 02139

William Rabinowitz, Cochair
Bose Corporation, The Mountain, Framingham, MA 01701

Chair's Introduction—1:35

Invited Papers

1:40

1pPPa1. In "Lou" of the Lou you know. Constantine Trahiotis (Neurosci., UConn Health Ctr., 6 Lyme Pl., Avon, CT 06001, tino@uchc.edu)

Lou Braida has distinguished himself as a scientist, educator, and, to the dismay of many, critic-par excellence. Others will speak to his wide variety of fundamental contributions to knowledge concerning several areas of auditory perception. I will confine my remarks to Lou the person and friend. In my interactions with Lou I discovered and have cherished an individual for whom many of you has remained "e-Lou-sive." I will introduce you to Lou the cheapskate, Lou the inventor, Lou the scammer, Lou the smuggler, Lou the travel agent, Lou the traveling chef, and Lou the practical engineer-observer. These other incarnations of Lou, taken together with the more widely known Lou, reveal the truly multidimensional ways in which he has led a very special life, one in which all of us have been fortunate to share.

2:00

1pPPa2. Cochlear mechanisms underlying the sharp frequency selectivity of hearing. Dennis M. Freeman, Roozbeh Ghaffari, Shirin Farrahi, and Jonathan B. Sellon (Res. Lab. of Electronics, Massachusetts Inst. of Technol., 77 Massachusetts Ave., MIT Rm. 7-133, Cambridge, MA 02139, freeman@mit.edu)

Sharp frequency selectivity, which is a hallmark of mammalian hearing, originates in the cochlea. However, the underlying mechanisms remain unclear. The pioneering work of von Bekesy showed that sounds launch waves of motion along the spiraling basilar membrane, and subsequent hydrodynamic analysis has shown how mechanical properties of the cochlear partition can interact with fluid forces to support sharp frequency tuning. These analyses have generally presumed (or even purported to prove) that longitudinal mechanical coupling through cochlear structures is negligible. Here, we demonstrate that the visco-elastic structure of the tectorial membrane (TM), a gelatinous structure that overlies the sensory receptor cells and plays a key role in stimulating them, also supports traveling waves. The distance over which TM waves propagate provides a measure of mechanical coupling and, through the cochlear map, determines a range of frequencies that correlates strikingly well with direct measurements of cochlear tuning in normal hearing mice, in mice with genetic disorders of hearing, and in humans. These results demonstrate significant longitudinal coupling through the TM and suggest that TM coupling plays an important role in determining the sharpness of cochlear frequency tuning.

2:20

1pPPa3. All I really need to know I learned from Lou and Nat: Lou Braida. Michael Picheny (Watson Multimodal, IBM TJ Watson Res. Ctr., POB 218, Yorktown Heights, NY 10598, picheny@us.ibm.com)

Lou Braida was one of my two primary mentors in graduate school at MIT. Lou taught me innumerable things. In this talk, I will only have time to focus on a few items. I will describe how I applied to Speech Recognition what I learned from him about the power of a psychophysical approach to research problems, the importance of good data collection, and the value of long-term spectral characteristics in perception. Speech recognition by now is a relatively mature field, but at the time the field was relatively unexplored territory. Lou's training allowed me to see ways to make advances in speech recognition experimental design, speaker-independent speech recognition, and noise-immune features and processing. While many of these early ideas have been subsumed over the years by more sophisticated processing, many of them have their roots in techniques inspired by a combination of perceptual knowledge with principled engineering design. Lou was, and is a master of both and I am forever grateful for his inspiration, mentoring, and friendship in shaping my career.

2:40

1pPPa4. Computational models of speech perception by cochlear implant users. Mario Svirsky and Elad Sagi (Otolaryngology-HNS, New York Univ., 550 First Ave., NBV-5E5, New York, NY 10010, mario.svirsky@nyumc.org)

Cochlear implant (CI) users have access to fewer acoustic cues than normal hearing listeners, resulting in less than perfect identification of phonemes (vowels and consonants), even in quiet. This makes it possible to develop models of phoneme identification based on CI users' ability to discriminate along a small set of linguistically-relevant continua. Vowel and consonant confusions made by CI users provide a very rich platform to test such models. The preliminary implementation of these models used a single perceptual dimension and was closely related to the model of intensity resolution developed jointly by Nat Durlach and Lou Braida. Extensions of this model to multiple dimensions, incorporating aspects of Lou's novel work on "crossmodal integration," have successfully explained patterns of vowel and consonant confusions; perception of "conflicting-cue" vowels; changes in vowel identification as a function of different intensity mapping curves and frequency-to-electrode maps; adaptation (or lack thereof) to changes in frequency-place functions; and some aspects of speech perception in noise. Our latest studies predict that enhanced phoneme identification by cochlear implant users may result from deactivation of a subset of electrodes in a patient's map. All these results build upon, and were made possible by concepts from Lou's work.

3:00–3:20 Break

3:20

1pPPa5. Early acoustic hearing and spoken language skills of children with cochlear implants. Rosalie M. Uchanski and Lisa S. Davidson (Otolaryngol., Washington Univ. in St Louis School of Medicine, 4523 Clayton Ave., Campus Box 8115, St. Louis, MO 63110, r.uchanski@wustl.edu)

Development of spoken language is difficult for children born with hearing loss. While most clinicians agree on the goal of improving the audibility of spoken language as early as possible, there is less agreement on the types of devices (bilateral cochlear implants vs. bilateral hearing aids vs. one hearing aid with one cochlear implant) they would recommend to achieve improved audibility. Additionally, acoustic properties of speech are conveyed differently by hearings aids (HAs) and cochlear implants (CIs); voice-pitch and prosodic properties, assumed critical for learning words from continuous speech, are conveyed better with HAs than CIs while broad spectral properties of individual speech segments are conveyed better with CIs than HAs. The relation between a simple model of a child's early (birth to ~3 years old) acoustic hearing experience (includes HA use, CI surgery dates, severity of hearing loss, etc.) and eventual spoken language skills (tested at later ages of 8-10 years old) will be examined, especially in the context of which devices might be best for spoken language development. This examination reflects Dr. Louis Braida's long-standing interest in understanding the acoustic properties of speech and its perception, especially for the benefit of those with hearing loss.

3:40

1pPPa6. Factors affecting accuracy and intelligibility of transliterators who use cued speech. Jean C. Krause (Commun. Sci. and Disord., Univ. of South Florida, 4202 E Fowler Ave., PCD 1017, Tampa, FL 33620, jeankrause@usf.edu)

Some deaf individuals access spoken information via transliterators who use Cued Speech, a system of hand gestures that supplement information available through speechreading alone. In this presentation, the accuracy and intelligibility of 12 transliterators with varying degrees of experience are examined. Accuracy, or the percentage of cues correctly produced, was evaluated at three different speaking rates (slow, normal, and fast), and intelligibility, or the percentage of words correctly received, was measured by presenting the materials that the transliterators produced to nine expert receivers of Cued Speech. Results show that speaking rate had a large negative effect on accuracy, caused primarily by an increase in omitted cues, while increased experience level was generally associated with increased accuracy. Intelligibility was generally higher than accuracy, with accuracy accounting for roughly 25% of the variance in intelligibility scores. We conclude by discussing factors such as speechreadability that could explain additional portions of the variance.

4:00

1pPPa7. Automated extraction of information from Language ENvironment Analysis (LENA) home recordings of older children with autism. Mark A. Clements, Rahul Pawar, Desmond Caulley (ECE, Georgia Inst. of Technol., School of ECE, Georgia Inst. of Technol., Atlanta, GA 30332-0250, clements@gatech.edu), Rebecca Jones, and Catherine Lord (Psychiatry, Weill Cornell Medicine, White Plains, NY)

It has been established that children and adolescents with Autism Spectrum Disorder show a wide range of abilities to use spoken words and establish interactive conversations. Automatic measurement of such abilities in naturalistic environments would greatly facilitate assessment and monitoring of such individuals. If sufficient accuracy could be achieved, studies could be performed whose sample sizes are large enough to draw meaningful conclusions. In the current study, 16-hour home audio recordings using the LENA (Language ENvironment Analysis) device are examined for older children and adolescents. Specific enhancements to the existing LENA analysis platform include the ability to diarize recordings for subjects aged 5 through 13, to detect non-verbal vocalizations such as laughter and whining, to identify child-directed speech, and to determine when questions are posed. Other higher-level descriptors involve extraction of affect, computation of conversational interaction measures, detection of cross-talk and interruption events, and identifying emotional outbursts. Subject-specific diarization based on a small amount of hand-labeled data yields acceptable accuracy. However, a newly developed system based on i-vectors, specifically designed for the environment at hand, requires no such labeling at the onset.

4:20

1pPPa8. Louis Braida's influence on speech intelligibility research. Karen Payton (Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747-2300, kpayton@umassd.edu)

In the Sensory Communications Laboratory at MIT, several researchers have grappled with the question of how to predict listeners' perceptual performance using quantitative metrics. Most of the work has been motivated by the goal of understanding the effect of acoustic degradations on speech intelligibility for hearing impaired listeners and determining the best way to mitigate or counter those degradations. Louis Braida has been interested in this topic for many years. He worked with Ken Grant to investigate augmentation of the Articulation Index (AI) with visual information to obtain an audio-visual AI. He continued this work with other students modeling perceptual integration across modalities. Lou and I worked together to investigate the ability of the Speech Transmission Index (STI) to predict intelligibility for impaired listeners. Initially, we tried to use it as a way to try and capture acoustic differences between conversational and clearly-articulated speech. That evolved into the development of a speech-based STI. Ray Goldsworthy extended our work, comparing several speech-based STI techniques and developing a new one to predict speech intelligibility for cochlear implant users. This talk will review the work done in this area under Lou's mentorship and recent advances.

4:40

1pPPa9. The auditory-visual articulation Index. Ken W. Grant (National Military Audiol. and Speech-Lang. Pathol. Ctr., Walter Reed National Military Medical Ctr., 301 Hamilton Ave., Silver Spring, MD 20901, ken.w.grant@gmail.com) and Joshua G. Bernstein (National Military Audiol. and Speech-Lang. Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD)

Hearing aids (HAs) are the primary method for treating hearing impairment. However, in complex environments with competing sound sources, HAs provide marginal benefits at best. Under these conditions, clinicians recommend facing the speaker to extract visual speech information. Combined auditory-visual (AV) speech generally provides a signal that is much more resistant to noise and reverberation than an auditory-only signal. The Articulation Index (AI) established that different frequency regions of speech vary in their degree of importance for intelligibility. However, frequencies most important for auditory-only speech intelligibility differ from those that are most important for AV speech intelligibility. Thus, the optimal signal-processing solution may differ between AV and auditory-only conditions. Braida and colleagues sought to develop an AV version of the AI to enable HA signal-processing strategies to be compared without the time and expense required for behavioral testing. This presentation describes this and other work inspired by Braida's efforts to predict AV speech intelligibility using only auditory-only and visual-only information. [Work supported by a grant from CDMRP, #DM130027. The views expressed in this abstract are those of the authors and do not reflect the official policy of the Department of Army/Navy/Air Force, Department of Defense, or U.S. Government.]

5:00

1pPPa10. Contributions of Louis Braida to improved signal processing for hearing aids: Addressing the problem of reduced dynamic range in listeners with sensorineural hearing loss. Charlotte M. Reed, Joseph G. Desloge, and Laura A. D'Aquila (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Rm. 36-751, MIT, 77 Massachusetts Ave., Cambridge, MA 02139, cmreed@mit.edu)

Lou's early work in the area of improved signal processing for hearing aids included his research on compression amplification to combat the effects of loudness recruitment in listeners with sensorineural hearing loss. Working with his doctoral students (including Rich Lippmann, Steve De Gennaro, and Diane Bustamante), Lou made major contributions towards an analytical understanding of the benefits and limitations of compression amplification as a component of hearing aids. Recently, Lou has been involved in work on a new signal-processing scheme which operates to equalize the energy in a speech signal over time. This energy-equalization (EEQ) scheme shares a similar goal with compression amplification in that they both attempt to match the range of speech levels into the reduced dynamic range of a listener with sensorineural loss. Their operation, however, is different: while compression amplification is based on the actual sound-pressure level of the signal, the EEQ scheme operates on relative energy calculations designed to reduce the variations in overall signal level. In this talk, we will describe the EEQ processing system together with results obtained on its evaluation with hearing-impaired listeners for speech reception in backgrounds of continuous and fluctuating noise.

Session 1pPPb

Psychological and Physiological Acoustics: Perception of Synthetic Sound Fields II

Sascha Spors, Cochair
Institute of Communications Engineering, University of Rostock, Richard-Wagner-Strasse 31, Rostock 18119, Germany

Nils Peters, Cochair
Advanced Tech R&D, Qualcomm Technologies, Inc., 5775 Morehouse Drive, San Diego, CA 92121

Contributed Papers

1:40

1pPPb1. Measuring speech intelligibility with speech and noise interferers in a loudspeaker-based virtual sound environment. Axel Ahrens, Márton Marschall, and Torsten Dau (Dept. of Elec. Eng., Hearing Systems group, Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, aahr@elektro.dtu.dk)

Loudspeaker-based virtual sound environments (VSEs) are emerging as a versatile tool for studying human auditory perception. In order to investigate the reproducibility of simple sound scenes, speech reception thresholds (SRTs) were measured with two interferers and in two spatial conditions (co-located and $\pm 30°$ separated) using the Danish matrix sentence test Dantale II (Wagener *et al*., 2003). SRTs were measured in a typical listening room and in a VSE consisting of a spherical 64-channel loudspeaker array using simulated room acoustics with mixed-order-Ambisonics (MOA) playback. The speech maskers were taken from the same material as the target (different talker, same sex). The noise maskers had the same long-term spectrum and broadband envelope as the speech interferer but had random phase (Best *et al*., 2013). The co-located conditions were reproduced comparably in the real room and in the VSE, with both speech and noise interferers. However, spatial separation led to a 3 dB higher benefit in the VSE than in the real room in both interferer conditions. Previous studies using a larger number of sound sources and more reverberation did not show such systematic differences between virtual and reference conditions, suggesting that reproduction errors may be masked in more complex scenes.

2:00

1pPPb2. Validating a perceptual distraction model in a personal two-zone sound system. Jussi Rämö, Lasse Christensen (Electron. Systems, Aalborg Univ., Fredrik Bajers Vej 7, Aalborg 9220, Denmark, jur@es.aau.dk), Søren Bech (Bang & Olufsen, Struer, Denmark), and Søren H. Jensen (Electron. Systems, Aalborg Univ., Aalborg, Denmark)

This paper focuses on validating a perceptual distraction model, which aims to predict user's perceived distraction caused by audio-on-audio interference, e.g., two competing audio sources within the same listening space. Originally, the distraction model was trained with music-on-music stimuli using a simple loudspeaker setup, consisting of only two loudspeakers, one for the target sound source and the other for the interfering sound source. Recently, the model was successfully validated in a complex personal sound-zone system with speech-on-music stimuli. Second round of validations were conducted by physically altering the sound-zone system and running a set of new listening experiments utilizing two sound zones within the sound-zone system. Thus, validating the model using a different sound-zone system with both speech-on-music and music-on-speech stimuli sets. Preliminary results show that the model performance is equally good in both zones, i.e., with both speech-on-music and music-on-speech stimuli, and comparable to the previous validation round (RMSE approximately 10%). The results further confirm that the distraction model can be used as a valuable tool in evaluating and optimizing the performance of personal sound-zone systems.

2:20

1pPPb3. The effect of reverberation and audio spatialization on egocentric distance estimation of objects in stereoscopic virtual reality. Will Bailey (Acoust. Res. Ctr., Univ. of Salford, Newton Bldg., Crescent, Salford M5 4WT, United Kingdom, j.w.bailey@edu.salford.ac.uk) and Bruno M. Fazenda (Acoust. Res. Ctr., Univ. of Salford, Manchester, United Kingdom)

It has been reported by numerous studies on distance perception in VR that a compression of visual space occurs in virtual environments presented using stereoscopic techniques. Other studies have shown that modified environmental auditory cues can affect egocentric spatial perception and that increased order of modality improved the experience of immersive media. Work was conducted to measure the effect of spatialized acoustic cues on egocentric distance estimation in head mounted display VR. Results suggest that the although early reflection content was not found to have significant effect on estimation of distance, presence of reverberation increases the perception of distance for objects further than 5m from the user and can compensate for the spatial compression observed in the use of stereoscopic VR.

Invited Papers

2:40

1pPPb4. Evaluation of techniques for navigation of higher-order ambisonics. Joseph G. Tylka and Edgar Choueiri (Mech. & Aerosp. Eng., Princeton Univ., MAE Dept. E-Quad, Olden St., Princeton, NJ 08544, josephgt@princeton.edu)

Metrics are presented that assess spectral coloration and localization errors incurred by navigational techniques for higher-order ambisonics. Previous studies on the coloration induced by such navigational techniques have been largely qualitative, and the accuracy of previously-used localization models in this context is unclear. The presented metrics are applied in numerical simulations of navigation over a range of translation distances. Coloration is predicted using an auditory filter bank to compute the spectral energy differences between the test and reference signals in critical bands, and localization is predicted using a precedence-effect-based localization model. Coloration and localization errors are also measured through corresponding binaural-synthesis-based listening tests, wherein subjects are first asked to rate the induced coloration relative to reference and low-pass-filtered "anchor" signals, and subsequently judge source position. Relationships are drawn between the metrics and the results of the listening tests in order to validate the predictive capabilities of the metrics.

3:00

1pPPb5. Perceptual evaluation of multichannel synthesis of moving sounds as a function of rendering strategies and velocity. Cédric Camier and Catherine Guastavino (Multimodal Interaction Lab, McGill Univ., 3661 McTavish St., Montreal, QC H3A 1X1, Canada, cedric.camier@gmail.com)

Sound-field synthesis for static sound sources has been extensively studied. Recently, dynamic sound sources synthesis has garnered increased attention. Classical sound-field rendering strategies discretize dynamic sound-fields as a sequence of stationary snapshots. The use of discrete multichannel arrays can generate further artifacts with moving sounds. Depending on the technique used, this results in an amplitude modulation due to successive loudspeaker contributions (VBAP) or in multiple comb-filtering (WFS) which could affect localization cues, especially at off-centered listening positions. We first present a detailed description of these artifacts. We then introduce a hybrid rendering strategy combining propagation simulation and VBAP at audio rate. We used this rendering strategy and WFS to synthesize white noise revolving around listeners on a circular 48-loudspeaker array. On each trial, participants had to identify the trajectory (circle, triangle, or square) for velocities ranging from 0.5 to 2 revolutions per second. Performance was well above chance level in all conditions. While WFS outperformed the hybrid rendering strategy at low velocities, no significant differences were observed at high velocities for which participants relied on temporal cues rather than spatial cues. The results highlight how artifacts of the rendering strategies interfere with dynamic sound localization at different velocities.

3:20–3:40 Break

Contributed Paper

3:40

1pPPb6. Determining the accuracy of sound field synthesis systems in reproducing subjective source locations. Anna C. Catton, Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincolc, 1110 S. 67th St., Omaha, NE 68182-0816, anna.catton@huskers.unl.edu), Adam K. Bosen, Timothy J. Vallier, and Douglas H. Keefe (Boys Town National Res. Hospital, Omaha, NE)

Sound field synthesis systems developed and applied to study human hearing perception differ in terms of the number and arrangement of loudspeakers in rooms of different sizes, and methods used to generate virtual sound environments. Research has evaluated how well such systems physically reproduce room acoustic conditions, but limited subjective data are available. This paper seeks to develop a method for determining the accuracy of the perceived localization of a virtual sound source generated using a multichannel sound synthesis system. A test method is applied to the sound field synthesis facility at Boys Town National Research Hospital, a room (5.8 m x 5.2 m x 2.7 m) with reverberation time of 0.16 s at 125 Hz and below 0.024 s at 250 Hz and above. Short bursts of broadband speech-shaped noise are presented at a number of virtual source locations under free-field and modeled reverberant-room conditions, and listeners are asked to point to the subjective source location. Subjective localization results are compared as functions of virtual sound location, and parameters of early reflections and reverberation in the modeled sound environment. Results are intended to guide future research on subjective room acoustics relevant to children's' communication needs. [Work supported by NIH GM109023.]

Invited Papers

4:00

1pPPb7. Contributions of head-related transfer function choice and head tracking to virtual loudspeaker binaural rendering. Brian F. Katz (Lutheries - Acoustique - Musique, Inst. ∂'Alembert, UPMC/CNRS, ∂'Alembert, boîte 162, 4 Pl. Jussieu, Paris 75252 Paris Cedex 05, France, brian.katz@upmc.fr), Peter Stitt, Laurent Simon (LIMSI, CNRS, Université Paris-Saclay, Orsay, France), Etienne Hendrickx (LABSTICC (Laboratoire des Sci. et techniques de l'information, de la Commun. et de la connaissance), Université de Bretagne Occidentale, Brest, France), and Areti Andreopoulou (LIMSI, CNRS, Université Paris-Saclay, Orsay, France)

This presentation will provide an overview of recent and ongoing studies regarding evaluations of sound fields using virtual loudspeaker binaural synthesis. Of specific interest is an identification of perceptual attributes affected by Head-Related Transfer Function (HRTF) choice beyond basic localization error and the sensitivity of listeners to head tracking with regards to latency and externalization judgments. A list of perceptual attributes, created using a Consensus Vocabulary Protocol elicitation method, and validated through listening tests, resulted in eight valid perceptual attributes for describing the perceptual dimensions affected by HRTF set variations. Employing prescribed head movements, sensitivity to head tracker latency showed small but significant differences between single and multichannel audio source scenes. A similar protocol was employed to comparing the sense of externalization as a function of head rotation with and without head tracking. In contrast to several previous studies, results showed that head movements can substantially enhance externalization, especially for frontal and rear sources, and that externalization can persist even though the subject has stopped moving his/her head. These works were carried out during the course of the French funded BiLi (Binaural Listening) project (FUI-AAP14).

4:20

1pPPb8. Auralization of acoustic spaces based on spherical microphone array recordings. Jens Ahrens (Chalmers Univ. of Technol., Sven Hultins gata 8A, Gothenburg 412 58, Sweden, jens.ahrens@chalmers.se), Christoph Hohnerlein (Technische Universität Berlin, Berlin, Germany), and Carl Andersson (Chalmers Univ. of Technol., Göteborg, Sweden)

Microphone arrays can capture the physical structure of a sound field. They are therefore potentially suited to capture and preserve the sound of acoustic spaces within given physical limitations that are determined by the construction of the array. Especially spherical microphone arrays have received considerable attention in this context. Superposed onto the limitations of the microphone array are the limitations caused by the auralization system. We present results from user studies on the perceptual differences between spherical microphone array recordings that are auralized with headphones as well as with a circular 56-channel loudspeaker array and headphone auralization based on dummy head measurements of the same spaces. Head-tracking was applied in all cases in which headphones were used.

4:40

1pPPb9. Sound environment and sound field reproduction using transducer arrays: Correlation between physical and perceptual evaluations. Philippe-Aubert Gauthier and Alain Berry (Mech. Eng., Université de Sherbrooke, 51, 8e Ave. Sud, Sherbrooke, QC J1G 2P6, Canada, philippe_aubert_gauthier@hotmail.com)

Sound Environment Reproduction (SER) using Sound Field Reproduction (SFR) is aimed at the spatial reconstruction of a target sound field captured using a microphone array. SFR has recently gained attention for SER in industrial or engineering contexts for sound comfort or sound quality studies. The challenge is to create a reproduced sound field that first satisfies an assessment based on physical evaluation, for example to satisfy any regulation based on physical quantities. However, the reproduced sound environment should also success in perceptual evaluation. In this work, SER was applied to spatial sound field simulation in a vehicle mock-up. Both physical and perceptual evaluations were completed. Physical metrics such as frequency-dependent averaged reproduction error (both phase and magnitude) and averaged magnitude error (ignoring phase) were measured. Perceptual evaluations were based on similarity listening tests while comparing SER with an original reference (the target sound field). Perceptual evaluations were compiled as similarity scores. Correlation of similarity scores based on various physical evaluations suggests that the frequency-averaged and spatially averaged magnitude error is the physical evaluation metric that is the most correlated with the result of listening tests. This suggests that spatially reproducing the accurate frequency spectrum is the first criterion for immersing SER.

Contributed Paper

5:00

1pPPb10. High-density data sonification of stock market information in an immersive virtual environment. Samuel Chabot and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 40 3rd St., Troy, NY 12180, chabos2@rpi.edu)

Data sonification is an important tool to enhance a user's ability to capture and process complex information. In this system, stock market data for the top 128 publicly traded stock options are analyzed for the sonification. This information includes the daily trading price and volume of each stock. Audio streams for conveying the daily information of each stock are generated using sine tone click trains, pitch alterations, and noise bursts. Each audio stream is mapped to an individual loudspeaker in the 128 loudspeaker array of Rensselaer's Collaborative-Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab) to create a high-density spatialized sonification within the immersive virtual environment. [Work supported by NSF #1229391 and the Cognitive and Immersive Systems Laboratory (CISL).]

Session 1pSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration I

Benjamin Shafer, Chair
Technical Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406

Contributed Papers

1:20

1pSA1. A study of complex system dynamics: Metronome synchronization. Noah A. Sonne and Teresa J. Ryan (Eng., East Carolina Univ., 200 Blue Beech Dr., Greenville, NC 27858, noahasonne@gmail.com)

This work investigates energy exchange within a complex vibrating system. The system is made up of a mass, called the primary oscillator, and a number of attached smaller structures, called the subordinate oscillator array. Specifically, a rectangular rigid foam base is the primary mass and mechanical metronomes are used as the subordinate oscillators. This work explores how the orientation and arrangement of the metronomes on the master structure affects the time it takes for metronome synchronization as well as the resulting amplitude of oscillation of the vibration of the primary mass. A MATLAB based image processing approach is used to measure these system parameter. [This work was supported by the Robert W. Young Award for Undergraduate Student Research.]

1:40

1pSA2. Vibration and sound of a flapping airfoil: The limit of small bending rigidity. Avshalom Manela and Michael Weidenfeld (Aerosp. Eng., Technion, Technion City, Haifa 32000, Israel, amanela@technion.ac.il)

We investigate the near and far fields of a thin elastic airfoil set at uniform low-Mach flow and subject to leading-edge heaving actuation. The airfoil is "hanged" in the vertical direction and is free at its downstream end, so that "hanging chain" gravity-induced tension forces apply. The structure bending rigidity is assumed small, and we focus on analyzing the differences between a highly elastic airfoil and a membrane (where the bending rigidity vanishes). The near field is studied based on potential thin airfoil theory, whereas the acoustic field is investigated using the Powell-Howe acoustic analogy. The results shed light on the specific effect of structure bending stiffness on the dynamics and acoustic disturbance of an airfoil.

2:00

1pSA3. Structural-acoustic optimization based on Fast Multipole Boundary Element Method sensitivity analysis of a coupled acoustic fluid-structure system. Nian Yang (State Key Lab. of Ocean Eng., Shanghai Jiao Tong Univ., Dongchuan 800 Rd., Shanghai, Shanghai 200240, China, yangnian@sjtu.edu.cn), Leilei Chen (College of Civil Eng., Xinyang Normal Univ., Xinyang, Henan, China), Kheirollah Sepahvand (Dept. of Mech. Eng.,Tech. University of Munich, Inst. of VibroAcoust. of Vehicles and Machines, Garching bei Munich, Germany), Hong Yi (State Key Lab. of Ocean Eng., Shanghai Jiao Tong Univ., Shanghai, China), and Steffen Marburg (Dept. of Mech. Eng.,Tech. University of Munich, Inst. of VibroAcoust. of Vehicles and Machines, Munich, Germany)

For the light structures immersed in water, a full fluid-structure interaction (FSI) has to be considered when structural acoustics is analyzed. However, the computation costs for the FSI prediction and optimization is always huge. The Fast Multipole Boundary Element Method (FMBEM) is one of the most widely used methods to accelerate the computations. Meanwhile, sensitivity analysis for FSI problems is the most time-consuming part of gradient-based optimization strategies. In this research, a FMBEM-based sensitivity analysis is applied to a practical underwater model's structural-acoustic optimization. Object function that represents the overall radiated sound power are investigated, where the damping material thickness in some specific areas were chosen as design parameters. An improvement of the object functions are found within a limited number of function evaluations. The FEM/FMBEM based sensitivity analysis is used to calculate the sensitivity of the object function to the design parameters. The method of moving asymptotes (MMA) is chosen as the optimization algorithm. The efficiency of this optimization strategy applied on FSI practical model is investigated in detail.

2:20

1pSA4. Ball sensor rattle: Experimental and numerical sensitivity study for seatbelt applications. Kai-Ulrich J. Machens, Jens Neumann, Jens Scholz (Occupant Safety Systems, ZF TRW Active & Passive Safety Technol., Industriestr. 20, Alfdorf D-73553, Germany, kai-ulrich.machens@zf.com), and Marian Markiewicz (Novicos GmbH, Hamburg, Germany)

Seatbelt systems are important elements of automotive safety systems. Most seatbelt retractors are equipped with ball sensors, enabling retractors to comply with mandatory vehicle sensitive locking requirements. The functional principle is based on ball inertia plus defined backlash of mating surfaces, which renders seatbelt retractors susceptible to rattle. Sensor ball rattle is considered as the most persistent parasitic noise source in occupant safety industry. The vibration induced noise behavior of the ball sensor is analyzed, both experimentally and by numerical simulation, predicting the sound pressure spectrum up to 5 kHz. Impact forces among mating surfaces are computed in the time domain with flexible multibody system analysis, employing Craig-Bampton modes to approximate the vibrations of acoustically relevant substructures. Acoustic radiation into the sound field is determined in the frequency domain using preprocessed acoustic transfer vectors from boundary element method analysis, significantly reducing computation time. A sensitivity study with variation of sensor mass, gap, and excitation demonstrates excellent correlation between numerical model prediction and experimental result across a large variety of test cases. The presented methodology can predict rattle induced noise, and therefore delivers substantial input for retractor design. Furthermore, other applications beyond seatbelt retractors could equally benefit from using this approach.

2:40

1pSA5. A methodology to design multi-axis test rigs for vibration and durability testing using frequency response functions. Polat Sendur (Mech. Eng., Ozyegin Univ., Nisantepe Mahallesi, Orman Sokak, Cekmekoy, Istanbul 34794, Turkey, polat.sendur@ozyegin.edu.tr), Umut Ozcan, and Berk Ozoguz (Ford Otomotiv Sanayi A.S, Istanbul, Turkey)

The multi-axis simulators are designed for experimental verification of the safe functioning of large components and subsystems under real world customer usage in vibration and durability testing. Transformation of the full vehicle conditions to mast rig testing with correct system dynamics and

vibration characteristics and boundary conditions is a key challenge in the development of the experimental set-up. In this paper, a systematic methodology is formalized how to design the experimental set-up on MAST rig to replicate the vehicle dynamics and vibration characteristics in vehicle conditions. System modes and frequency response functions are chosen as key performance metrics to compare the dynamics of the system to be tested for both full vehicle and rig design. Criteria on the metrics are defined to make decision if the test rig design is sufficiently replicating the in-vehicle conditions. The methodology is illustrated on a side skirt attached to a heavy duty truck chassis that demonstrates the application of the methodology in practice.

3:00–3:20 Break

3:20

1pSA6. Structural health monitoring under random flow loading. Nicola Roveri, Silvia Milana, Antonio Culla, and Antonio Carcaterra (Dept. of Mech. and Aerosp. Eng., Univ. of Rome La Sapienza, via Eudossiana 18, Rome, Italy, nicola.roveri@gmail.com)

The aim of the work is the analysis of fluid-structure systems, when excited by a flow consisting of an incompressible potential fluid with embedded vortexes. In fact, in many problems of relevant application interests, the monitoring and the potentially detection of structural damages in structures undergoing loads in operative conditions is important. The present method tries to identify simultaneously the load characteristics together with the structural damage. The flow is characterized by the average velocity of the fluid conveying the vortexes and by the position and intensity of the conveyed vortexes. A method for the identification of these flow parameters, based on vibration signals measured at the elastic fluid-structure interface, is proposed. Vibration signals are numerically generated and then processed with time-frequency techniques, such as the ensemble empirical mode decomposition and the normalized Hilbert transform. The sensitivity of the algorithm to the measurement position and to single versus multi-points acquisitions are also investigated. A particular instantaneous frequency is firstly employed to estimate the load characteristics. The influence of the load is then removed from the instantaneous frequency, so that the damage position can be identified. The validity of the proposed method is analyzed varying the flow parameters, the damage locations and depths; effect of ambient noise is also taken into account.

3:40

1pSA7. Design of in-plane functionally graded material plates for improved vibration characteristics. Nabeel T. Alshabatat (Mech. Eng., Tafila Tech. Univ., Tafila, Jordan), Kyle R. Myers (Penn State Univ., Appl. Res. Lab., 3220B G Thomas Water Tunl, PO Box 30, State College, PA 16804, krm25@arl.psu.edu), and Koorosh Naghshineh (Mech. & Aerosp. Eng., Western Michigan Univ., Kalamazoo, MI)

A method for improving the vibration characteristics of plate structures is proposed. This method uses functionally graded material (FGM) instead of isotropic material to construct the plates. The volume fraction of each material constituent is defined in the plane of the plate by a 2D trigonometric law, while the material properties through the thickness are assumed constant. The finite element method is used for modal and harmonic analysis, and a genetic algorithm is utilized for optimization of the chosen objective function. The efficacy of the method is demonstrated by two design problems. In the first design problem, FGM is used to maximize the fundamental frequencies of plates with different boundary conditions. In the second design problem, the kinetic energy of a vibrating FGM plate is minimized at a specific excitation frequency. These example design problems show that material tailoring of plate structures using FGM can result in substantial improvements of their vibration characteristics. The results can be used to guide the practical design of FGM plates to enhance their dynamic properties.

4:00

1pSA8. Structural-acoustic optimization using cluster computing. Robert Campbell, Micah R. Shepherd, and Stephen Hambric (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, rlcampbell@psu.edu)

Structural-acoustic optimization using state-of-the-art evolutionary algorithms may require tens of thousands of system solutions, which can be time-limiting for full-scale systems. To reduce the time required for each function evaluation, parallel processing techniques are used to solve the system in a highly-scalable fashion. The system acoustic radiation is modeled as a stochastic problem using finite elements for the structural vibration and boundary elements for the fluid loading and acoustic analysis. The approach is demonstrated by minimizing the sound radiated from curved panel under the influence of a turbulent boundary layer in the presence of added point masses. Details of the point mass magnitudes and distribution are outcomes of the optimization. Solver scaling information is provided that demonstrates the utility of the parallel processing approach.

4:20

1pSA9. Acoustic testing techniques for replicating in-flight dynamic loads. Kobi J. Cohen and Daniella Raveh (Aerosp. Eng., Technion, 3 Harottem St., Apt. 37, Haifa 3584706, Israel, kobic8@gmail.com)

Modern weapon systems used on combat aircrafts have complex electronic assemblies that are required to operate in challenging dynamic environment throughout their life cycle. Among the various sources of excitation, aerodynamic noise is considered most significant. The paper presents vibroacoustic measurements from captive flight, and attempts to replicate them in acoustic laboratory testing. The question of interest is which testing method, in terms of configuration and control scheme, is the most adequate to accurately simulate the vibratory response of inner assemblies to flight loads. The paper examines acoustic test methods in a reverberant chamber. The tested article is a subsystem of a weapon system that includes electrical assemblies, integrated inside a structural envelope. Two test configurations are compared—"covered," in which the subsystem is tested inside its structural envelope, and "uncovered," in which the subsystem is directly exposed to acoustic excitation. Acceleration measurements show that when excited by in-flight acoustic levels, the acceleration responses of the uncovered subsystem, are significantly lower than those measured in flight. For the covered configuration, although the acoustic levels inside the envelope are attenuated by the structure, the resulting accelerations are significantly higher and closer to those of flight.

4:40

1pSA10. Dissipation as energy transport to molecular scale. Adnan Akay (Bilkent Univ., Ankara 06800, Turkey, akay@cmu.edu)

Dissipation describes irreversible transfer of ordered kinetic energy from a larger-scale to thermalized vibrations at the molecular scale. Considering dissipation as transfer of energy from one form of vibrations to another form, models can be developed without the need for qualitative empirical constants. Examples of "lossless" damping mechanisms will be derived to illustrate energy conversion at the molecular scale.

Session 1pSC

Speech Communication: Non-Native Speech and Bilingualism (Poster Session)

Kristin Van Engen, Chair
Washington University in St. Louis, One Brookings Dr., Campus Box 1125, Saint Louis, MO 63130-4899

All posters will be on display from 1:20 p.m. to 5:20 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 1:20 p.m. to 3:20 p.m. and authors of even-numbered papers will be at their posters from 3:20 p.m. to 5:20 p.m.

Contributed Papers

1pSC1. Acoustical analysis of English /r/ and /l/ by native Japanese adults and children. Katsura Aoyama (Audiol. and Speech-Lang. Pathol., Univ. of North Texas, 1155 Union Circle #305010, Denton, TX 76209, katsura.aoyama@unt.edu), James E. Flege (Univ. of Alabama at Birmingham, Tuscania, Italy), Reiko Akahane-Yamada (ATR, Seika cho, Kyoto, Japan), and Tsuneo Yamada (Open Univ. of Japan, Chiba, Japan)

This study investigated the acoustic properties of American English /r/ and /l/ produced by native Japanese (NJ) and native English (NE) speakers. The purpose of this study was to examine the differences in production reported in Aoyama *et al.* (2004) acoustically. Aoyama *et al.* evaluated productions of /r/ and /l/ in 64 NE and NJ adults and children (16 participants each in 4 groups) using intelligibility ratings. The data were collected twice to study the acquisition of English by the NJ adults and children. In this study, four acoustic parameters (duration, F1, F2, and F3) were measured in 256 tokens each of /r/ and /l/. The results showed that all of the acoustic parameters differed significantly between NJ and NE speakers at both times of testing. Some aspects of acoustic parameters changed significantly over the course of one year in NJ children's productions. Lastly, the formant values in NJ speakers' productions indicated that their productions of both /r/ and /l/ resembled NE speakers' productions of /r/ more than the NE speakers' productions of /l/. This finding is consistent with Aoyama *et al.*'s claim that NJ speakers may have more difficulty in producing English /l/ than with English /r/.

1pSC2. Factors influencing intelligibility and fluency in non-native speech. Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, mbaesebe@uoregon.edu) and Tuuli Morrill (George Mason Univ., Fairfax, VA)

Substantial research has examined the factors making non-native speech more difficult to understand than native speech. Prior work has suggested that speaking rate is one such factor, with slower speech being perceived as less comprehensible and more accented. Further, non-native speech is produced with shorter utterances and more frequent pauses than native speech. Recent work has suggested that in addition to non-native speech being produced more slowly than native speech, it is produced with a more variable speaking rate. In the present study, we examine the relationship between variability in speaking rate, pausing and utterance length, and intelligibility and fluency ratings of non-native speech. We asked listeners to transcribe sentences produced by non-native speakers and to rate the fluency of read speech. Preliminary results suggest that rate variability does correlate with intelligibility of non-native speech, but not of native speech, and rate variability does not correlate as strongly with fluency. In addition, pause duration may interact with sentence complexity, but appears independent of rate. These results suggest that while fluency and intelligibility are abstract constructs, examining variability in non-native speech and its relationship to a number of factors may help explain why non-native speech is difficult to understand.

1pSC3. Individual variation in the perception of different types of speech degradation. Drew J. McLaughlin, Melissa M. Baese-Berk (Linguist, Univ. of Oregon, 1290, Eugene, OR 97403, dmclaug2@uoregon.edu), Tessa Bent (Speech and Hearing Sci., Indiana Univ., Bloomington, IN), Stephanie A. Borrie (Communicative Disord. and Deaf Education, Utah State Univ., Logan, UT), and Kristin Van Engen (Psychol. and Brain Sci., Washington Univ., Saint Louis, MO)

Both environmental noise and talker-related variation (e.g., accented speech) can create adverse listening conditions for speech communication. Individuals recruit additional cognitive, linguistic, or perceptual resources when faced with such challenges, and they vary in their ability to understand degraded speech. However, it is unclear whether listeners employ the same additional resources when encountering different types of challenging listening conditions. In the present study, we compare individuals' ability on a variety of skills —including vocabulary, selective attention, rhythm perception, and working memory—with transcription accuracy (i.e., intelligibility scores) of speech degraded by the addition of speech-shaped noise or multi-talker babble and/or talker variation (i.e., a non-native speaker). Initial analyses show that intelligibility scores across degradations of the same class (i.e., either environmental or talker-related) significantly correlate, but correlations of intelligibility scores across degradation classes are weaker. The relationship between intelligibility scores and cognitive-linguistic skills is similar, showing that while vocabulary and working memory correlate with multiple degradation types, rhythm perception only correlates with environmental degradations. Taken together, these results indicate that listeners may recruit different resources when faced with different classes of listening challenges.

1pSC4. Effects of talker intelligibility and noise on judgments of accentedness. Sarah Gittleman (Washington Univ. in St. Louis, 6023 Waterman Blvd., 2W, St. Louis, MO 63112, sgittleman@wustl.edu) and Kristin Van Engen (Washington Univ. in St. Louis, Saint Louis, MO)

The damaging effect of background noise on the intelligibility of foreign-accented speech has been well documented, but little is known about the effect of noise on listeners' subjective judgments of accents. Noise adds distortion to speech, which may cause it to sound more "foreign." On the other hand, noise may reduce perceived foreignness by masking cues to the accent. In this study, 40 native English speakers listened to 14 English-speaking native Mandarin speakers in four levels of noise: -4 dB, 0 dB, +4 dB, and quiet. Participants judged each speaker on a scale from 1 (native-like) to 9 (foreign). The results showed a significant decrease in perceived accentedness as noise level increased. They also showed a significant interaction between noise and intelligibility: intelligibility (which had been measured for the same talkers in a previous study) had the greatest effect on perceived accentedness in quiet, and a reduced effect with increasing noise levels. These findings indicate that listeners' decreased access to acoustic-phonetic cues in the presence of background noise also reduces their sensitivity to phonetic variation arising from foreign accents. Furthermore, the link between intelligibility and accentedness is weakened by the presence of noise.

1pSC5. Early and late Spanish-English bilingual adults' identification of American English vowels. Miriam Baigorri (Long Island Univ. Brooklyn, 1 University Plaza, Brooklyn, NY 11201, miriam.baigorri@liu.edu) and Erika S Levy (Teachers College, Columbia Univ., New York, NY)

Increasing numbers of Hispanic immigrants are entering the US (US Census Bureau, 2011) and are learning American English (AE) as a second language (L2). Accurate perception of AE vowels is important because vowels carry a large part of the speech signal (Kewley-Port, Burkle, & Lee, 2007). The present study examined the accuracy with which early and late Spanish-English bilingual adults identify AE vowels. Listeners were presented with AE vowels (/i/, / I /, /ɛ/, /ʌ/, /æ /, /ɑ/, and /o/) in a /gəbVpə/ context. They were instructed to click on the key word response from a choice of nonsense words that contained the second vowel they heard. Findings indicate that identification accuracy of L2 vowels was significantly higher with early age of L2 acquisition. However, early bilingual listeners' vowel perception was not native-like, suggesting that the phonetic properties of their native language influenced L2 speech perception. Additionally, identification accuracy varied as a function of the particular vowel, shedding light on how the relationship between Spanish and AE vowel inventories might explain the difficulties that arise in Spanish-English bilinguals' identification of AE vowels. Findings are examined in relation to perceptual assimilation and discrimination of the stimuli by the same listeners.

1pSC6. Second language pronunciation training using acoustic-to-articulatory inversion. Jeffrey J. Berry, Abigail Stoll (Speech Pathol. & Audiol., Marquette Univ., P.O. Box 1881, Milwaukee, WI 53201-1881, jeffrey.berry@marquette.edu), Deriq Jones, Seyedramin Alikiaamiri (Elec. and Comput. Eng., Marquette Univ., Milwaukee, WI), and Michael T. Johnson (Elec. and Comput. Eng., Univ. of Kentucky, Lexington, KY)

The current work presents articulatory kinematic, acoustic, and perceptual data characterizing the effects of how visual biofeedback derived from acoustic-to-articulatory inversion may influence vowel pronunciation training for native-Mandarin speakers of English. Ten participants were engaged in a six-week pronunciation training program that included a focus on English vowel production. As an addition to traditional pronunciation training techniques often used by speech-language pathologists, half of the participants were also provided with visual biofeedback displays detailing aspects of their current tongue position as well as idealized positions for vowel targets. Visual displays were obtained using an acoustic-to-articulatory inversion model based on the Parallel Reference Speaker Weighting (PRSW) method for model adaptation. Pre- and post-training changes in articulation were compared between participants that used only traditional pronunciation training methods and those who were given visual biofeedback based on acoustic-to-articulatory inversion. Data analyses focus on articulatory-kinematic measures, obtained via electromagnetic articulography, measures of vowel formant frequencies, and perceptual assessments based on phonetic transcriptions from expert listeners. The results of the current work provide insights regarding the value of PRSW-based adaptation of acoustic-to-articulatory inversion models and the resulting visual feedback displays as tools in second-language learning of English pronunciation.

1pSC7. Perception of native language speech sounds does not predict non-native speech sound learning. Pamela Fuhrmeister (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., U-1085, Storrs, CT 06269, pamela.fuhrmeister@uconn.edu) and Emily Myers (Speech, Lang., and Hearing Sci., Univ. of Connecticut, Storrs Mansfield, CT)

Individual differences are often observed in laboratory studies of non-native speech sound learning. One possible explanation for this variability is that better detection of fine-grained contrasts within *native* language categories might facilitate *non-native* learning. For instance, Diaz *et al*. (2008) found larger MMN responses to both native and non-native speech contrasts in good compared to poor perceivers of a non-native contrast, suggesting a general speech-related skill. The current study explores whether the ability to discriminate subtle differences in native language speech sounds correlates with non-native speech sound learning. To test this, we trained participants on a non-native, Hindi dental/retroflex contrast and assessed their categorization and discrimination of a native /da/-/ta/ continuum. Additionally, participants completed a visual Flanker task in order to control for general motivation in the experimental setting. Neither native language measures nor the Flanker task predicted non-native speech sound learning abilities. Rather, using k-means cluster analysis, we found two distinct groups of learners and non-learners that did not significantly differ on native language or Flanker measures, suggesting that non-native speech sound learning may be independent of those skills. Instead, non-native learning success was best predicted by an ability to discriminate the non-native contrast at pretest.

1pSC8. Perception and production of American English consonants /v/ and /w/ by Hindi speakers of English. Vikas Grover (Commun. Disord. and Deafness Dept., Kean Univ., NJ 07083, vgrover@kean.edu), Valerie Shafer, D. H. Whalen (Speech-Language-Hearing Sci., The Graduate Ctr., CUNY, New York, NY), and Erika Levy (Commun. Sci. and Disord., Teachers College, Columbia Univ., New York, NY)

This study examined the ability of Hindi speakers of English to perceive and produce American English (AE) consonants /v/ and /w/, which are difficult for Hindi speakers to distinguish (e.g., in "vest" and "west"). It also examined whether the Hindi listeners' length of residence (LOR) in the US affected their performance. Two groups of Hindi speakers were included: Hindi speakers who had been in the US for more than 5 years and Hindi speakers who lived in India and used English as their second language. Participants performed perception and production tasks of naturally produced tokens of word forms containing /v/ and /w/. Hindi listeners performed significantly less accurately than the English listeners on all tasks. The non-significant differences between the two Hindi groups indicated that the Hindi US groups' experience with the /v/-/w/ contrast in the US was insufficient to allow for perceptual learning of this contrast. The findings shed light on speech perception, production and comprehension (for lexical items that differ minimally, e.g., 'viper vs. wiper') challenges faced by native Hindi speakers learning English. This information can also be helpful for designing perception and production training programs for this population.

1pSC9. Relation between acoustic-phonetic properties and speech intelligibility in noise obtained with bilingual talkers. Sabine Hochmuth, Tim Jürgens, Thomas Brand, and Birger Kollmeier (Medical Phys. and Cluster of Excellence Hearing4all, Universität Oldenburg, Carl-von-Ossietzky Str. 9-11, Oldenburg 26129, Germany, sabine.hochmuth@uni-oldenburg.de)

An objective, language-independent way of predicting observed differences in speech intelligibility in noise across talkers based on their acoustic-phonetic properties was pursued by exploiting speech intelligibility data in stationary speech-shaped noise uttered by bilingual talkers and comparing inter-individual as well as intra-individual speech feature variations across languages. Matrix sentence materials were used that were uttered by bilingual talkers of German/Spanish and of German/Russian and by the respective original matrix test talkers. Various acoustic-phonetic parameters discussed in the literature as being related to speech intelligibility were determined for each talker. Vowel space area, between-vowel category dispersion, and energy in the mid-frequency region represented by the speech intelligibility index were found to be language-independent acoustic-phonetic properties most strongly related to speech intelligibility at least for German, Russian and Spanish. Generally larger inter-individual variation within languages than intra-individual variation across languages was found. Hence, objective phonetic criteria like vowel space area may be used in the future to objectively assess the potential of a talker to be easily understood in a noisy background. One reason of the generally poorer intelligibility performance of Spanish compared to German or Russian may lie in the usage of considerably smaller vowel space areas.

1pSC10. Interaction of drift and distinctiveness in L1 English-L2 Japanese learners. Marie K. Huffman (Dept. of Linguist, Stony Brook Univ., SBS S 201, Stony Brook, NY 11794-4376, marie.huffman@stonybrook.edu), Katharina Schuhmann (Dept. of Germanic and Slavic Lang. and Literatures, Penn State Univ., State College, PA), Kayla Keller, and Chanda Chen (Dept. of Linguist, Stony Brook Univ., Stony Brook, NY)

Dynamic L2 effects on L1 phonetics appear in experienced and novice second language learners, raising the question of what linguistic and cognitive factors determine their occurrence, degree, and direction (assimilatory versus dissimilatory). Unlike Chang (2012), our longitudinal data from voiceless stops in early L1 English:L2 Japanese learners show primarily dissimilatory increase in English VOTs, an effect found most strongly early in their first semester. Assimilatory L1 drift toward the lower VOTs of L2 Japanese may be disfavored because a decrease in English voiceless stop VOT could threaten the L1 contrast between long and short lag stops. Furthermore, dissimilatory VOT increase on voiceless stops allows English speakers to distinguish phonetically similar L1 and L2 voiceless stops (e.g., Flege and Eefting 1987). These two principles predict that our voiced stop data could show increased L1 prevoicing (an assimilatory effect that would not endanger the English voicing contrast), while also displaying non-identical prevoicing/short-lag values for L1 and L2 voiced stops (separating the languages, as for Huffman and Schuhmann's (2016) English-Spanish learners). Overall, our data suggest that L1 changes in early L2 learning can be dissimilatory, and that phonetic properties of L1 and L2 contrasts affect how L1 values restructure during early L2 acquisition.

1pSC11. The effect of second language orthographic input on the learning of Mandarin words. Yen-Chen Hao (Modern Foreign Lang. and Literatures, Univ. of Tennessee, Knoxville, TN) and Chung-Lin Yang (Psychol. and Brain Sci., Indiana Univ., Memorial Hall 322, 1021 E 3rd St, Bloomington, IN 47408, cy1@indiana.edu)

This study examines the effect of L2 orthography on Mandarin word learning. English speakers at three proficiency levels participated in a Mandarin word-learning experiment. During the learning phase, half of the participants were provided with Pinyin (Chinese Romanization) and tone marks (the PY group), while the other half were provided with characters (the CH group). After learning, the participants judged the matching of sound and meaning of 64 pairs, half of which were matches, while the other half were either segmental or tonal-mismatch items. The results showed that the Advanced and Intermediate learners in the CH group were more accurate than their counterparts in the PY group in the tonal-mismatch and match conditions respectively. In contrast, the naïve participants in the PY group were more accurate with the matches than those in the CH group. Also, participants in the PY group were overall inaccurate with the tonal-mismatch items regardless of their proficiency levels. However, in the CH group the Advanced learners scored significantly higher with the tonal-mismatch items than the other groups. This study suggests that characters are more effective than Pinyin in helping L2 learners encode the sounds of new Mandarin words, especially in tone encoding.

1pSC12. Is it *blow* or *below*? Non-native listeners' perception of words that contrast in syllable count. Keiichi Tajima (Dept. of Psych., Hosei Univ., 2-17-1 Fujimi, Chiyoda-ku, Tokyo 102-8160, Japan, tajima@hosei.ac.jp) and Stefanie Shattuck-Hufnagel (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA)

Non-native speakers often have difficulty accurately producing and perceiving the syllable structure of a second language. For example, Japanese learners of English often insert epenthetic vowels when producing English words, e.g., *stress* produced as /sutoresu/. Similarly, when asked to count syllables in spoken English words, they frequently overestimate the number of syllables, suggesting that they tend to perceptually insert epenthetic vowels between adjacent consonants. These tendencies suggest the possibility that learners may have difficulty distinguishing between English words that contrast in syllable count, i.e., words that differ in the presence/absence of a vowel, e.g., *blow-below*, *sport-support*. Furthermore, if listeners perceptually insert epenthetic vowels, then they should misperceive *blow* as *below* more often than *below* as *blow*. To test these predictions, Japanese listeners participated in a 2AFC identification task, using 78 English minimal pairs contrasting in syllable count such as *blow-below*. Results showed that Japanese listeners indeed had difficulty with this task. However, misidentification of *blow*-type words as *below* was *less* frequent than misidentification of *below*-type words as *blow*, contrary to predictions based on perceptual epenthesis. These results suggest that simple comparison of syllable structure between languages may not suffice to predict difficulties in L2 speech perception. [Work supported by JSPS.]

1pSC13. Costs and cues in code-switched lexical access. Alice Shen (Dept. of Linguist, Univ. of California Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94704, azshen@berkeley.edu)

While perceiving code-switches incurs processing costs [Soares & Grosjean (1984, Memory & Cognition 12(4):380-386)], bilingual listeners use acoustic cues to anticipate switches and facilitate processing [Fricke et al. (2016, J. Memory Lang. 89:110-137)]. This study investigates the role of anticipatory prosodic cues in the online processing of a code-switch from a non-tonal to a tonal language. Experiment 1 compares reaction times for perceiving Mandarin and English target words in English frames, to test whether recognizing code-switches is costly. Although reaction times for code-switched stimuli (475 ms) were not significantly slower than monolingual stimuli (466 ms), all participants self-reported as Mandarin-dominant speakers who frequently code-switch, suggesting that language dominance and experience may modulate switch costs. Reaction times to naturally-produced code-switched stimuli (426 ms) were faster than to spliced code-switched stimuli (511 ms), suggesting that anticipatory cues facilitate the perception of code-switches. Experiment 2 is a visual world eye-tracking task. The proportion of looks toward images corresponding to the target word (e.g., [mɑʊ̯^{4}tsɨ^5] "hat"), a cross-language phonetic competitor (e.g., [maʊs] "mouse"), and a within-language phonetic competitor (e.g., [mɑʊ̯^{2}tɕin^1] "towel") are compared to assess any influence of anticipatory cues on target and non-target language activation levels during auditory recognition of code-switches.

1pSC14. Assimilatory and dissimilatory L1 English vowel drift in early learners of Japanese. Katharina Schuhmann (Dept. of Germanic and Slavic Lang. and Literatures, Penn State Univ., State College, PA, Katharina.Schuhmann@gmail.com) and Marie K. Huffman (Dept. of Linguist, Stony Brook Univ., Stony Brook, NY)

In contrast to predictions of the Speech Learning Model, Chang's (2012) study of novice English-speaking learners of Korean finds no clear segment level assimilation between phonetically similar English and Korean vowels. The relative complexity of the Korean and English vowel systems may have made L2 to L1 vowel association inconsistent across speakers, leading to contradictory segment level effects. We examined vowels in L1 English:L2 Japanese learners, hypothesizing that the less dense vowel space in Japanese would simplify L1:L2 segment associations. Specifically, English [ɑ] and Japanese [a] should be associated by learners in a way that could lead to segment level L1 drift effects, allowing us to determine whether assimilatory L1 drift would occur in English for novice learners of Japanese, as Flege (1987, 1995) and Chang predict. Formant data for students in their first and second semester of Japanese instruction show mostly L1 assimilatory drift in F1, and some L1 dissimilation in F2, with most speakers making one change but not both. Overall, English [ɑ] variants stay within L1 norms, highlighting the importance of L1 phonetic repertoire in constraining L1 drift effects. Results will be compared to data for [i] and [u] to determine whether systemic level drift also occurs.

1pSC15. Top-down influence on phonetic categorization of native vs. non-native speech. Jessamyn L. Schertz (Dept. of Lang. Studies, Univ. of Toronto Mississauga, 1265 Military Trail, Humanities Wing, HW427, Toronto, ON M1C 1A4, Canada, jessamyn.schertz@utoronto.ca) and Kara E. Hawthorne (Dept. of Commun. Sci. and Disord., Univ. of MS, Edmonton, AB, Canada)

Speech perception requires integration of multiple sources of information, including bottom-up acoustic information and top-down contextual information, and listeners may adjust their reliance on a given source of

information depending on the communicative context. This work tests the hypothesis that listeners increase reliance on contextual, relative to acoustic, information when listening to a talker with a foreign accent, under the assumption that the bottom-up information (non-native pronunciation) may be less reliable. Native English listeners categorized an utterance-final target word, where the initial consonant systematically varied in voice onset time (VOT), as either "goat" or "coat." Target words were embedded in carrier sentences contextually biased towards one of the words (e.g., "The girl milked the [coat/goat]" vs. "The girl put on her [coat/goat]"). Stimuli were created from productions by two talkers: a native English talker and a native Mandarin/L2 English talker with a discernable foreign accent. As expected, acoustic information (VOT) was the primary cue for categorization, but sentence context also influenced perception in both talker conditions. Furthermore, preliminary results indicate that the semantic context effect is larger in the Accented than in the Native condition, suggesting that listeners do indeed increase reliance on contextual information when listening to foreign-accented speech.

1pSC16. Task differences do not impede overall learning when adapting to a novel sound. David Saltzman and Emily B. Myers (Speech, Lang., & Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT 06269, david.saltzman@uconn.edu)

Lexically guided perceptual learning (LGPL) and second-language learning (L2) research seeks to understand how listeners adapt to novel speech sounds. In LGPL, listeners shift their perception of a native phonetic contrast in response to hearing an ambiguous token embedded in an unambiguous lexical context. In L2 tasks, often involve categorizing non-native sounds, usually with explicit feedback. In general, L2 learning is seen as more effortful and individually variable than LGPL. However, paradigms differ in terms of stimuli (native vs. non-native phonetic contrasts) as well as tasks (lexically-guided implicit feedback vs. explicit feedback). To test whether L2-type vs. LGPL-type tasks yield differences in learning, participants were trained using an L2-style task to shift their native boundary along an /s/ to /sh/ continuum. With feedback, participants categorized the midpoint token as one novel object, and an endpoint token (counterbalanced across subjects) as another novel object. After training, a boundary shift comparable to previous studies using the LGPL task was found, suggesting that explicit and lexically-guided feedback produce similar magnitude of learning. Despite this, participants in the L2-style task experienced lower accuracy and more variability in perception of the continuum endpoints, suggesting that task differences partially explain less consistency across individuals in L2 research.

1pSC17. Perception of Russian palatalization contrasts by English listeners. Kevin Roon (CUNY Graduate Ctr., 365 Fifth Ave., Ste. 7107, New York, NY 10013, kroon@gc.cuny.edu) and D. H. Whalen (Haskins Labs., New Haven, CT)

Russian contrasts palatalized vs. non-palatalized consonants across primary oral articulator, manner, voicing, and word position, in both stressed and unstressed syllables. This palatalization contrast is challenging for native English speakers to master, possibly due to English speakers not being able to discriminate the relevant linguistic contrast in all the environments in which it exists in Russian. Previous studies have shown that English listeners are good at discriminating this Russian contrast prevocalically, but there is no experimental evidence indicating how well they discriminate this contrast in the wide variety of environments in which it is used in Russian, and when produced by different talkers. The present study tested how well English listeners could perceive the Russian palatalization contrast across manner, word position, primary oral articulator, and talker. 24 listeners performed an AX discrimination task in which A and X were always produced by two different speakers, one male and one female. Trials on which A and X mismatched differed in palatalization (e.g., /p/-/p^{j}/) or manner (e.g., /t/ vs. /s/). The results show the combinations of environments that present the greatest challenges for English listeners in discriminating the Russian contrast, with syllable-final obstruents being especially hard.

1pSC18. Temporal-order processing of American-English vowel sequences by native and non-native English-speaking listeners. Catherine L. Rogers, Bogyeong Cheon, and Gail Donaldson (Dept. of Commun. Sci. and Disord., Univ. of South Florida, USF, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

To understand the development of native-like proficiency in speech processing, we must consider the apparent ease with which native speakers process speech sounds under a variety of conditions. In the present study, auditory temporal-order processing of American-English vowel sequences was compared across three listener groups: monolingual English speakers and relatively early vs. later learners of English as a second language. Using the methods of Fogerty, Humes and Kewley-Port [2010, *J. Acoust. Soc. Am., 127,* 2509-2520], 70-ms resynthesized versions of the syllables "pit, pet, put," and "pot" were presented in a two-syllable temporal-order processing task. Task difficulty was increased by decreasing syllable-onset asynchrony (SOA), i.e., the duration between syllable onsets. SOA thresholds for accuracy of syllable-sequence identification were estimated using the method of constant stimuli on each of four 72-trial blocks. Similar SOA thresholds were obtained for native English speakers and early learners of English, but SOA thresholds increased by a factor of two or more for later learners of English. Furthermore, the average SOA threshold of the later learners is similar to that of the older listeners in Fogerty *et al.* (2010), suggesting that increased processing time partially accounts for both groups' increased difficulty in processing speech in noisy environments.

1pSC19. Perceptual similarity spaces of British English vowels by speakers of Pakistani Urdu. Ishrat Rehman and Amalia Arvaniti (English Lang. and Linguist, Univ. of Kent, SECL, Canterbury, Kent CT2 7NF, United Kingdom, I.Rehman@kent.ac.uk)

Free classification was undertaken with 70 listeners from Lahore, Pakistan with Punjabi and Urdu as their first languages in order to shed light on the English vowel features that have been most relevant in developing the "new English" variety spoken by them (*Pakistani English*). The stimuli were 19 *hVd* words carrying the Southern British English (SSBE) vowels. The responses were statistically analyzed using hierarchical clustering and multidimensional scaling. Listeners were sensitive to both F1 and F2, but could not distinguish the high-mid from the low-mid vowels. The central vowels /ɜː/ and /ʌ/ were in different groups indicating greater sensitivity to backness (F2): listeners classed /ʌ/ as back, but /ɜː/, which is more fronted in SSBE, as front. Diphthongs were grouped with monophthongs, sometimes based on the initial, sometimes on the final element; /aʊ/ /aɪ/ and /ɔɪ/, however, formed a separate group, possibly because their first and last element are most distinct. Listeners were not sensitive to duration: /iː/~/ɪ/, /ʊ/~/uː/, and /ɔː/~/ɒ/ were grouped with each other. Overall, the results indicate that Punjabi-Urdu listeners are sensitive to both height and backness, but possibly prioritize the latter, while they lack an intermediate (central) space, all features consistent with characteristics of Pakistani English.

1pSC20. Perceptual assimilation of Mandarin Chinese consonants by native Danish listeners. Sidsel Rasmussen (English, Aarhus Univ., Jens Chr. Skous Vej 4, Århus C 8000, Denmark, sira@cc.au.dk) and Ocke-Schwen Bohn (English, Aarhus Univ., Aarhus, Denmark)

A perceptual assimilation experiment examined the cross-linguistic mapping of Mandarin Chinese initial consonants to Danish consonants. 24 native Danish listeners were auditorily presented with naturally produced CV syllables which consisted of the Mandarin initial consonants [p, p^{h}, t, t^{h}, k, k^{h}, x, ts, ts^{h}, s, tɕ, tɕ^{h}, ɕ, tʂ, tʂ^{h} ʂ, ɻ, w, j] and vowels from the set [a, u, i, y] so that none of the CV syllables violated the permissible combinations of C and V in Mandarin. The Danish listeners identified the initial consonant of the stimuli with phonetically unambiguous Danish orthographic symbols and provided goodness ratings for each match. We found that assimilation patterns for Mandarin consonants differed greatly as a function of the following vowel, suggesting that native expectations regarding coarticulatory effects of V on C importantly affect perceptual assimilation. For example, Mandarin [tɕ, tɕ^{h}, ɕ] were assimilated to Danish <dj, tj, sj>, respectively, when followed by [a, y], but to Danish <t, t, s> when followed by [i]. The detailed results allow us to generate precise predictions for the discriminability and thus learnability of Mandarin consonants for native Danish listeners and learners.

1pSC21. Interdental-stopping among older-aged bilingual and monolingual Finnish- and Italian-heritage Upper Peninsula Michiganders. Paige Cornillie, Julianne Fosgard, Samantha Gibbs, Delani Griffin, Olivia Lawson, and Wil A. Rankinen (Commun. Sci. and Disord., Grand Valley State Univ., 515 Michigan St. NE, Ste. 300, Office 309, Grand Rapids, MI 49503, wil.rankinen@gvsu.edu)

The realization of interdental fricatives as coronal oral stops, referred to as interdental-stopping, is often attributed to substrate effects of ethnicity and immigrant-heritage. Michigan's Upper Peninsula (UP) is an excellent case to examine this feature's complex variation within bilingual and monolingual older-aged cross-sections of a rural immigrant speech community. To what degree, if any, is interdental-stopping occurring among Michigan's UP Finnish and Italian-heritage speech communities? Interdental-stopping has been documented in UP English [3, 2], but only a more recent report has provided any quantitative account that tracks this salient feature and its sociophonetic trends within the community [1]. The present study examines this feature among 41 Finnish-Americans and 30 Italian-Americans, whom are all older-aged residents from Michigan's Marquette County. Both samples are stratified by gender, socioeconomic status and lingua-dominance. All data is obtained from a reading passage task. This study reveals interdental-stopping occurring most often among Italian working-class males and least among English-dominant bilinguals. Such findings goes beyond the claim that this feature of UP English indexes working-class [2]. Interdental-stopping indexes working-class masculinity—held with prestige and used by older-aged, working-class males, primarily among Italians, as a linguistic marker of local identity in the UP English speech community.

1pSC22. How within-category gradience in lexical tones influences native Chinese listeners and second-language Chinese learners recognize words: An eye-tracking study. Zhen Qin, Jie Zhang, and Annie Tremblay (Linguist, Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045, qinzhenquentin2@ku.edu)

This study investigates whether within-category gradience in lexical tones influences native and non-native Chinese listeners' word recognition. Previous offline research found that Chinese listeners have a more categorical perception of lexical tones, and thus show less sensitivity to within-category variability in tones, than non-native listeners. However, it is unclear whether native and non-native listeners have sensitivity to within-category gradience during online word recognition. Native Chinese listeners and proficient adult English-speaking Chinese learners were tested in a visual-world eye-tracking experiment. The target was a level tone and the competitor was a high-rising tone, or vice versa. The auditory stimuli were manipulated such that the target tone was either canonical in the standard condition, phonetically more distant from the competitor in the distant condition, or phonetically closer to the competitor in the close condition. Growth curve analysis on fixations suggested that native listeners showed a gradient pattern of lexical competition, with less competition in the distant condition and more competition in the close condition than in the standard condition; learners, on the other hand, showed increased competition in both the distant and close conditions than in the standard condition. The native and non-native listeners' difference suggested the influence of their language backgrounds.

1pSC23. Can second language suprasegmentals be learned? A study on Japanese learners of Italian as foreign language. Elisa Pellegrino (Univ. of Zurich, Plattenstrasse 54, Zuerich 8032, Switzerland, pellegrino.elisa.1981@gmail.com)

One of the biggest challenges for L2 learners is to develop native-like prosodic competence. There are evidence that computer-assisted pronunciation training helps learners perceive and produce L2 suprasegmentals. In this study we test to the effectiveness of one of the techniques developed in the area of spoken language technology for education and language learning—self-imitation—to facilitate the acquisition of the rhythmic and prosodic characteristics of Italian. 7 Japanese learners of Italian as Foreign language (NNSs) and 2 Italian native speakers (NSs) were asked to read aloud and record two sentences in Italian conveying different pragmatic functions. The utterances of NNSs were manipulated as they receive the segmental durational characteristics as well as the f0 characteristics of the corresponding NSs' utterances. NNSs imitated their own voice previously modified to match the reference NSs and recorded the new performance. To quantify the degree of approximation to the native prosodic and rhythmic pattern after self-imitation, pre- and post-training utterances were compared to those of the native models by speech rate, f0, and segmental durational characteristics. Preliminary results show that after the training utterance duration and vocalic intervals length better match the target duration.

1pSC24. Effects of experience on the production of Korean stop consonant contrasts by Mandarin Chinese learners. Grace E. Oh (English Lit. and Lang., Konkuk Univ., 120 Neungdong-ro, Gwangjin-gu, Seoul 05029, Korea (the Republic of), gracey1980@yahoo.com)

The effect of L2 experience on the segmental and prosodic production of second language was investigated. Thirty two Chinese learners of Korean varying in the amount of experienced (3 months vs. 2 years) were compared to sixteen age-matched native Korean speakers in their production of three-way contrastive stops (aspirated, lenis, and fortis) in Korean. To examine both segmental and prosodic aspects, Korean four-syllable phrases (i.e., Accentual Phrase) beginning with each stop type in word-initial position were elicited. VOT, *f0*, H1-H2 were analyzed and compared across groups (native Korean, experienced, inexperienced groups). For further analyses of the prosodic domain, *f0* of the first two syllables (High-High for aspirated, fortis and Low-High for lenis) was compared. The results revealed that the experienced Chinese learners showed an early mastery of fortis stops, producing more native-like VOT and H1-H2 than the inexperienced group. Also, Chinese groups were able to produce a sequence of AP with a native-like *f0* pattern regardless of the amount of experience. The production of *f0* for stop contrasts, on the other hand, was non-native like even after 2 years of experience, supporting previous observations that segments require greater L2 experience than intonation to be acquired in a native-like manner.

1pSC25. Identification of vowels of two different varieties of English by native speakers of Japanese and Korean. Takeshi Nozawa (Lang. Education Ctr., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, t-nozawa@ec.ritsumei.ac.jp) and Sang Yee Cheon (Dept. of East Asian Lang. and Literatures, Univ. of Hawaii, Honolulu, HI)

Native speakers of Japanese and Korean heard and identified /i, ɪ, eɪ, ɛ, æ, ɑ, ʌ/ uttered in /bVd/, /dVd/ and /kVd/ frames by native speakers of American and New Zealand English. New Zealand English has gone through an idiosyncratic vowel shift. For instance, /æ/ and /ɛ/ are raised and /ɪ/ is centralized. Overall American English vowels are identified more accurately by the two listener groups. Both listener groups identified American English /ɛ, æ/ better than New Zealand English equivalents, but on the contrary American English /ɑ/ is less accurately identified than New Zealand English /ɑ/ (or /ɒ/). Despite these similarities, some differences are observed between the two listener groups. While Japanese listeners identified New Zealand English /i, ɪ,/ less accurately than American English /i, ɪ/, Korean listeners identified New Zealand English /ɪ/ more accurately than American English /ɪ/. Japanese listeners outperformed Korean listeners in identifying American English /eɪ/ and New Zealand English /ɑ,/, but Korean listeners identified American English /ʌ/ and New Zealand English /ɪ/, /æ/ and /ʌ/ better than did Japanese listeners. The results point to the effect of L1 phonology and the differences in what each listener group believe each English vowel sounds like. For instance, Korean non-high front vowels are acoustically more similar to New Zealand English vowels /æ/ than to American English corresponding vowel. [Work partially supported by Grant-in-Aid for Scientific Research (C)16K02650.]

1pSC26. Development of semantic context facilitation for nonnative-accented speech. Katherine E. Miller, Rachael F. Holt (Dept. of Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, miller.7940@osu.edu), Tessa Bent (Speech and Hearing Sci., Indiana Univ., Bloomington, IN), and Andrew Blank (Dept. of Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

Children can use sentence context to facilitate their understanding of both native and nonnative speakers and this benefit increases with age. In

this study, 5- to 7-year-old children (n = 90) and adults (n = 30) were compared on their ability to benefit from sentence context. In addition, we examined whether receptive vocabulary accounts for variability in spoken word recognition differently for words spoken in sentences versus in isolation. Stimuli were produced by either native- or nonnative-accented (Japanese and Spanish) speakers. Listeners first identified words in isolation that had been extracted from meaningful sentences. Then they completed the NIH Toolbox Picture Vocabulary Test. Finally, they identified the same spoken words embedded in the original sentences. Children and adults showed significant word recognition advantages for the sentence condition compared to the isolated word condition for both native and nonnative speakers. However, adults showed a much larger benefit from sentence context than the children in the nonnative but not the native condition. Further, older children benefited from context more than younger children. Finally, receptive vocabulary was positively correlated with nonnative recognition of words in sentences but not in isolation for both children and adults, suggesting that better receptive vocabulary enhances use of context for nonnative speech.

1pSC27. Improving speech recognition in noise through speaking style modifications for native and non-native listeners. Kirsten Meemann and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, 305 E. 23rd St., Mail Code B5100, Austin, TX 78712, kirsten.meemann@utexas.edu)

It is well established that native listeners outperform non-native listeners on word recognition tasks involving both speech-shaped noise (SSN) and competing speech (speech babble). The present study examined whether this non-native disadvantage can be compensated for by speaking style enhancements. We also explored how these acoustic-articulatory modifications interact with energetic and informational masking at different signal-to-noise ratios to determine intelligibility for the two listener groups. Native and non-native participants heard noise-adapted (NAS) and clear speech (CL) sentences mixed with either SSN, two-talker (2T), or six-talker (6T) babble. CL and NAS significantly improved word recognition in noise, but native listeners were better able to use the intelligibility-enhancing modifications. Results revealed an interaction between noise type and SNR such that the intelligibility gain was larger for SSN at an easier SNR, but for 6T babble at a harder SNR. The speaking style modifications enhanced intelligibility least in 2T babble for both listener groups. Speaking style adaptations improve word recognition under energetic masking (SSN), but are most beneficial when informational and energetic masking are combined (6T babble) and presented at a low SNR. The intelligibility benefit was smallest in listening conditions with less energetic masking (2T babble) that resulted from larger spectro-temporal dips.

1pSC28. Non-native word recognition in babble and white noise. Bin Li (Linguist and Translation, City Univ. of Hong Kong, 83 Tat Chee Ave., Kowloon Tong 000, Hong Kong, binli2@cityu.edu.hk)

Noise causes degradation in speech signals, which poses difficulties for non-native perception. In this study, we examined impacts of noisy conditions on word recognition in naturally produced sentences by Chinese speakers of English (CE) and native speakers of English (NE). We also manipulated the linguistic contexts where targeted words occur, in order to assess how syntactic and semantic information may contribute to facilitate speech perception in adverse conditions. We recorded and compared the mean accuracy of keyword rewriting, to examine effects of noise and linguistic cues across listener groups. Results show that babble noise had more severe influence on CE groups' performance, which was also found commensurate with their English proficiencies. NE listeners, however, showed different patterns in their tolerance of noise and in use of linguistic cues.

1pSC29. Overweighting of pitch cues in stop identification by Korean learners of Mandarin Chinese. Sang-Im Lee-Kim (Dept of Foreign Lang. and Literatures, National Chiao Tung Univ., Hsinchu, Taiwan, sangim119@gmail.com)

The present study reports on a novel case where learners of a tonal language not only develop keen sensitivity to F0 cues in general, but the increased sensitivity may also foster perceptual reorganization of cue weighting in stop identification. Korean learners of Mandarin and novice listeners participated in identification tasks for which pitch contours of Mandarin words containing unaspirated stops were digitally manipulated. In word-initial position, learners showed considerably higher sensitivity to onset pitch cues, showing a near-categorical perception from lenis to fortis judgment along with higher onset pitch. In word-medial position, tone contours emerged as a significant predictor of the stops; predominant fortis judgment in high-level tone vs. lenis judgment in mid-rising tone. The novice listeners showed similar patterns in both cases, but to a much smaller degree. The effect of tone contours was discussed with reference to an auditory contrast whereby onset pitch of mid-rising tone is perceptually lower, providing positive evidence for lenis stops, due to the contrast with the following high pitch. Taken together, the learners' differential behavior suggests substantial reorganization of perceptual cues whereby onset pitch is promoted to a primary cue for the lenis-fortis contrast, concurrent with significant underweighting of VOT cues.

1pSC30. The effect of noise and second language on turn taking in task-oriented dialog. A Josefine Sørensen, Michal Fereczkowski, and Ewen N. MacDonald (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ctr. for Hearing and Speech Sci., Bldg. 352, Ørsteds Plads, Kgs Lyngby DK-2800, Denmark, emcd@elektro.dtu.dk)

Previous studies of floor transfer offsets (FTO), the interval between one talker stopping and the other starting, suggest that normal conversation requires interlocutors to predict when each other will finish their turn. We hypothesized that noise and/or speaking in a second language (L2) would result in longer FTOs due to increased processing demands. Conversations from 20 pairs of normal hearing, native-Danish talkers were elicited using the Diapix task in four conditions consisting of combinations of language (Danish vs. English) and noise background (quiet vs. ICRA 7 noise presented at 70 dBA). Overall, participants took longer to complete the task in both noise and in L2 indicating that both factors reduced communication efficiency. However, L2 had very little effect beyond completion time, likely because the participants were very good in English. In contrast to our predictions, in the presence of noise, the median of the FTO distribution decreased by approximately 30ms and the standard deviation decreased by approximately 10%. However, the average duration of interpausal units (i.e., utterances of continuous speech) increased by 40% in noise. These findings are consistent with talkers holding their turn for longer, allowing more time for speech planning.

1pSC31. The power of a unimodal distribution in cue reweighting: Unimodality vs prediction error as signs of cue irrelevance. Zara Harmon (Linguist, Univ. of Oregon, Eugene, OR), Kaori Idemaru (East Asian Lang. & Literatures, Univ. of Oregon, Eugene, OR 97403, idemaru@uoregon.edu), and Vsevolod Kapatsinski (Linguist, Univ. of Oregon, Eugene, OR)

Maye & Gerken (2000) proposed that sound categories can be learned from probability distributions: a unimodal distribution suggests a single category, while a bimodal one suggests two contrasting ones. Research on distributional learning has focused on developing a contrast through exposure to a bimodal distribution. Here, we instead investigate how exposure to a unimodal distribution affects perception of a pre-existing multidimensional contrast (voicing, for which the primary cue is VOT). A total of 60 adult native English speakers were exposed to either bimodal or unimodal VOT distributions spanning the unaspirated/aspirated boundary (*bear*/*pear*). However, we paired acoustic stimuli with pictures of bears and pears independently of VOT in training. For each stimulus, participants were asked to guess the referent and received (random) feedback, generating an error signal that suggested VOT is no longer informative and should be downweighed. In this design, the bimodal distribution suggests the existence of two categories but provides a clearer error signal: in the unimodal condition, most training tokens have ambiguous VOT, preventing clear predictions of voicing, thereby reducing prediction error. Nonetheless, participants downweighed VOT (and upweighed a secondary cue, F0) only with unimodal training. We conclude that unimodality is a very strong cue to dimensional irrelevance.

1pSC32. Do linguistic and cognitive processing differences impact vocoded speech understanding? Arifi N. Waked and Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, 0100 Lefrak Hall, College Park, MD 20740, awaked@umd.edu)

Normal-hearing individuals presented vocoded speech show great variability, even after explicit training. Assuming similar neural encoding, this variability might be explained by linguistic and cognitive factors. Therefore, we measured vocoded speech understanding in participants that should show a range of linguistic and cognitive abilities; namely, children (8-10 years) and adults ($\geq$18 years), and monolingual English speakers and bilingual Spanish-English speakers. We hypothesized that bilingual adults have a "cognitive advantage" in their understanding of vocoded speech, related to reinforcement of executive function skills by using multiple languages. We also hypothesized that children may rely relatively more on their linguistic knowledge, as they may not have developed these same cognitive skills and strategies. Participants were trained on speech understanding simulating shallow cochlear implant insertion depth (6-mm frequency-to-place mismatch) with auditory and visual feedback. Between training trials, participants were tested without feedback on standard (0-mm) or shallow (6-mm) simulated insertion depths. Participants were then tested on measures of phonological awareness and vocabulary in English and/or Spanish and the five cognitive measures of the NIH Cognitive Toolbox. Associations between these measures and speech perception may allow us to better tailor therapy methods to members of particular age/language groups. [Work supported by NIH grant R01AG051603.]

1pSC33. Vowel undershoot in the production of nonwords by English and Mandarin speakers. Chung-Lin Yang (Psychol. & Brain Sci., Indiana Univ., 1101 E 10th St., Bloomington, IN 47405, cy1@indiana.edu), Yu-Jung Lin, and Kuan-Yi Chao (Linguist, Indiana Univ., Bloomington, IN)

In a previous study (Yang, 2011), it was found that both Mandarin and English speakers showed vowel undershoot (characterized by decreased vowel duration and lowered first formant) in the production of English vowels while reading a list of real words, but Mandarin speakers could not make a clear tense-lax distinction. The current study aims at examining the degree of undershoot and tense-lax distinction when speakers had to reproduce what they heard without any visual input. In a nonword repetition task, Mandarin and English speakers were auditorily presented with a list, consisting of 8 English nonword triplets with the target vowels /i, ɪ, eɪ, ɛ/ embedded: monosyllables, disyllables and trisyllables. Each trial began with a short training session where participants listened to the nonwords in each triplet as many times as they wanted. After training, each nonword were played three times and participants repeated after each token. Our preliminary data showed that English speakers, while maintaining the tense-lax distinction, showed undershoot when producing nonwords across the three syllabic conditions. On the contrary, Mandarin speakers showed very limited undershoot and unclear tense-lax distinction across the three syllabic conditions, especially between /eɪ/ and /ɛ/. [Yu-Jung Lin and Kuan-Yi Chao contributed equally to this project.]

1pSC34. Effects of linguistic experience on brainstem encoding of speech sounds. Tian Zhao and Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Box 367988, Seattle, WA 98195, zhaotc@uw.edu)

Linguistic experience has been demonstrated repeatedly over the past decades as an important factor influencing the perception of speech sounds. At the behavioral level, speakers of different languages identify and discriminate speech sounds differentially (e.g., categorical perception). This has been observed at the cortical level as a reduced Mismatch Response, a measure that can reflect the sensitivity of the cortex to the differences between sounds. Yet, the effect of linguistic experience at an earlier stage of speech processing, namely, the brainstem, is scarcely studied. In the current study, we therefore aimed to examine specifically whether linguistic experience affects speech sound processing at the brainstem level. Twenty native Spanish speakers and twenty monolingual English speakers are recruited and their brainstem response to speech sounds will be measured with EEG. Specifically, we will use synthesized speech sounds: /ba/ with a + 10 ms and a -40 ms voice-onset-time (VOT). We hypothesize that while the encoding by the two groups will be similar for the + 10 ms /ba/ (common to English and Spanish), English speakers' encoding of -40 ms prevoiced /ba/s (Spanish only) will be reduced relative to Spanish speakers. Data will be analyzed and interpreted with regards to these hypotheses.

1pSC35. Acoustic comparison on syllabic rates between stress-timed and syllable-timed language speakers. Myungsook Kim (English, Soongsil Univ., Sangdo-ro 369, Seoul 06978, South Korea, kimm@ssu.ac.kr) and myungjin bae (Dept. of IT Eng., Soongsil Univ., Seoul, South Korea)

Syllabic rates (i.e., phonation rate for each syllable) of a speaker provide us with a lot of interesting information about the language in question. Especially between syllable-timed language and stress-timed language, we may expect that speakers will show a gap in syllabic rates as they have their own ways to pronounce. Syllabic rates are closely related with information density in speech communication although the Rosetta Project (2011) found a negative correlation between the two illustrating the existence of encoding strategies for each of 7 popular languages in the world. Since Korean was not on the list for the Project, this paper further investigates and compares acoustic characteristics on syllabic rates between two speakers, one of English, stress-timed language, and the other of Korean, syllable-timed language, when speaking both Korean and English. The result shows that the English speaker speaks Korean 1.39 times faster than the Korean speaker in syllabic rate. The English speaker also shows higher energy distribution in high frequency range while the Korean speaker shows a relatively even distribution in all frequency ranges. When speaking English, the Korean speaker speaks slower than the English speaker, showing not much difference in duration, pitch, and intensity between the stressed and unstressed syllables. The paper concludes that keeping a proper syllabic rate for a target language might be important in language acquisition.

1pSC36. Analysis of allophones based on audio signal recordings and parameterization. Andrzej Czyżewski, Magdalena Piotrowska, and Bozena Kostek (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, andczyz@gmail.com)

The aim of this study is to develop an allophonic description of English plosive consonants based on recordings of 600 specially selected words. Allophonic variations addressed in the study may have two sources: positional and contextual. The former one depends on the syllabic or prosodic position in which a particular phoneme occurs. Contextual allophony is conditioned by the local phonetic environment. Co-articulation overlapping in time demands a precise determination of allophonic pronunciation in the context of phonemic transcription. The presented study is focused on creation of speech recordings that may serve for the analysis of allophone variation. Two sets of recordings are prepared. The first one consists of words read by the non-native speakers. Tempo of reading is forced by a teleprompter. In the second case, every word is played back from the recordings of the phonology expert and then the speaker repeats a particular word. The last stage is the assessment of recordings by the same expert. Scores assigned by the expert are included as a reference for signal analysis and parametrization. [Research sponsored by the Polish National Science Centre, Dec. No.2015/17/B/ST6/01874.]

Session 1pSP

Signal Processing in Acoustics: Application of Bayesian Methods to Acoustic Model Identification and Classification II

Edmund Sullivan, Cochair
Research, Prometheus, 46 Lawton Brook Lane, Portsmouth, RI 02871

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

Chair's Introduction—1:15

Invited Papers

1:20

1pSP1. Bayesian classification of environmental noise sources. Edward T. Nykaza, Matthew G. Blevins (ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61822, edward.t.nykaza@usace.army.mil), Carl R. Hart (ERDC-CRREL, Hanover, NH), and Anton Netchaev (ERDC-ITL, Vicksburg, MS)

Classification algorithms are an essential component of continuously running environmental noise monitors. Without them, one does not know which noise sources are responsible for the levels recorded by the monitor. This is problematic given that continuously recording monitors may accumulate millions of triggered events and terabytes of data. In this study, we look at the utility of Bayesian classification methods. We compare the performance of these methods to some of the top performing environmental noise classifiers (e.g., support vector machines, random forest, and bagged trees), and discuss the advantages and disadvantages of the Bayesian approach. In particular, we compare the accuracy, number of observations needed to achieve an accurate classification, computation time, and feature importance.

1:40

1pSP2. Bayesian model selection for multilayer microperforated panel sound absorber design. Cameron J. Fackler, Yiqiao Hou, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY, cfackler@gmail.com)

Microperforated panel (MPP) sound absorbers are capable of providing high sound absorption coefficients without the use of fibrous materials; however, they typically function in narrow frequency ranges. By combining multiple MPPs into a multilayer absorber, the frequency bandwidth may be increased while maintaining a high absorption coefficient. Modeling the acoustic properties of an MPP absorber requires four physical parameters per MPP layer. Since each additional MPP layer in a multilayer absorber increases the complexity of the acoustic model, Bayesian model selection is well-suited to the task of designing a multilayer MPP absorber. In such a design, minimizing the number of layers used while still satisfying the design goals is desirable, in order to optimize material usage, cost, and space required by the absorber. In a full Bayesian design framework, model selection determines the number of MPP layers required, while parameter estimation determines the (physical) design parameters for each layer. In this work, an example design scheme is specified to satisfy a practical need for acoustic absorption. The Bayesian framework produces a three-layer MPP design which meets the target requirements. This absorber design is constructed, and impedance tube measurements are obtained to validate the acoustic absorption properties.

2:00

1pSP3. Minimum mean square error estimation of a sparse broadband acoustic response with a hierarchical mixture Gaussian prior. Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, pgendron@umassd.edu) and Jacob L. Silva (ECE Dept., Univ. of Massachusetts Dartmouth, Darmouth, MA)

Considered here is a minimum mean square error (MMSE) estimator of a broadband acoustic response function over a vertical aperture based on an adaptive sparsity prior. The prior is a hierarchical Gaussian mixture distribution built on the assumption that acoustic paths can be partitioned into a relatively coherent set of arrivals that on average exhibit Doppler spreading about a mean rate and a set of incoherent paths that exhibit a flat Doppler spectra. The hierarchy establishes constraints on the parameters of each of these Gaussian models such that coherent components of the response are both sparse and in the ensemble obey the Doppler spread profile. An empirical Bayes approach is developed to estimate the latent parameters of the hierarchy, from which the shared time varying dilation process can ameliorated thereby enhancing coherent multi-path combining. The model is tested with acoustic communication recordings taken in shallow water at very low signal-to-noise ratios.

2:20

1pSP4. Speaker localization in a reverberant environment using spherical statistical modeling. Boaz Rafaely (Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, POB 653, Beer Sheva 84105, Israel, br@bgu.ac.il), Christopher Schymura, and Dorothea Kolossa (Ruhr-Universität Bochum, Bochum, Germany)

Estimation of the direction of arrival (DoA) of speakers in reverberant environments is an important audio signal processing task in a wide range of applications. Recently, a reverberation-robust method for DoA estimation has been developed. It is based on the identification of time-frequency bins that are dominated by the direct path from the source. As the DoA statistics was found to have a multimodal distribution, clustering using Gaussian mixture modeling improved localization accuracy. However, this method employed linear statistics over the azimuth and elevation angles, therefore introducing between 0 and 360 degrees azimuth. This work explores the use of spherical statistics in the direct-path dominance approach to speaker localization in reverberant rooms, providing a more suitable angular representation. Theoretical and experimental aspects of the proposed approach are investigated and validated using a computer simulation of a speaker in a room.

Contributed Papers

2:40

1pSP5. Inference on speed, depth, and range of a submerged object from a limited vertical aperture under uncertain noise power. Abner C. Barros (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Darmouth, MA 02747, abarros1@umassd.edu), David C. Anchieta (Elec. and Comput., Universidade Federal do Pará, Belém, Para, Brazil), and Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, North Dartmouth, MA)

Considered here is an narrow band directed source and hydrophone receiver arrangement employed to infer the depth, speed, and range of an oncoming submerged object. Tracking the scattering body by means of a continuous wave transmission is challenging due to the difficulty of inferring the frequencies and angles of the two returned closely spaced wave vectors. Computation of the posterior pdf of these two wave vectors is accomplished by a judicious Gibbs sampling scheme that accounts for the uncertainty in the ambient acoustic noise level. Computational improvements are accomplished by taking full advantage of the prior distribution of the wave vectors associated with the specific target scenario. Very short duration observations of approximately 10 milliseconds are considered over which the Doppler rate of change of the two wave vectors can be considered negligible. This Bayesian scheme takes advantage of the analytic tractability of the conditional density of the received amplitudes and phases and of the noise powers. The conditional densities of the ordered wave vectors however are constructed numerically by 2 dimensional inverse quantile sampling. The inferred joint density of depth, range, and speed of the target is accomplished by constructing an inverse transformation of the acoustic propagation model.

3:00

1pSP6. Machine learning applied to estimating broadband source signature characteristics in a shallow ocean environment. David P. Knobles (KSA, LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com) and Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE)

Probabilistic machine and deep learning methods are critical elements in the area of automation and artificial intelligence. The regression task considered here is to use supervised machine learning to predict a frequency dependent acoustic source level from received broadband signals that have propagated through a shallow ocean waveguide possessing random features. The details are intimately linked to a previously introduced sequential maximum entropy—Bayesian approach employed to generate marginal probability distributions for both environmental and source parameter values. To meet the requirement to have both low training and generalization errors, several regularization and non-linear optimization methods are considered to enhance performance. This includes convolutional networks. Optimization includes not only the issue of sampling but also finding an effective model capacity, which is one of the most significant challenges to successful machine learning applications. The methodology is applied to measured broadband data collected in about 75 m of water about 60 miles south of Cape Cod Massachusetts in an area called the mud pond. Both the training samples and the samples not used in training have known source levels with which to measure both the training error and the generalization error and thus quantify performance.

3:20–3:40 Break

3:40

1pSP7. Clustering technology for the analysis and classification of bioacoustic vocalizations. Ian Agranat (Wildlife Acoust., Inc., 3 Mill and Main Pl., Maynard, MA 01754, ian@wildlifeacoustics.com)

A new technique for clustering and classifying bioacoustics vocalizations from large audio recording data is presented. Over 30,000 terrestrial passive audio and ultrasonic recorders deployed worldwide for the monitoring of birds, frogs and bats generate many petabytes of data annually. Our methods seek to efficiently and automatically detect candidate vocalizations from these massive datasets and sort them into clusters of similar vocalization patterns. Researchers can then quickly survey biodiversity and find species of interest by examining each of the resulting clusters. After clustering, species-specific classifiers can be built by labeling whole clusters or training classifiers after labeling individual detections. Once a classifier is built, it can be used to search new data for species of interest. The algorithms detect vocalizations comprising one or more short syllables. A variable-length observation sequence is extracted comprising spectral features changing through time. A Hidden Markov Model is trained on these sequences and Fisher scores are calculated for each detection. The dot product of two Fisher scores provides a measure of similarity between two vocalizations and forms the basis for clustering. The algorithms are designed to run in parallel on multi-core processors and stream the data to minimize memory requirements.

4:00

1pSP8. Information loss due to environmental variability and uncertainty in Bayesian localization of a narrowband source. Thomas J. Hayward (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Performance degradation of Bayesian localization of a low-frequency narrowband acoustic source due to variability and imperfect knowledge of the acoustic environment is investigated in a computational study. The environmental variability is modeled as arising from water column fluctuations associated with a diffuse random linear internal wave field in a shallow-water ocean waveguide. The ambient noise spatial cross-spectrum is represented by a Kuperman-Ingenito model. For the case of complex Gaussian internal wave spectral amplitudes, a closed-form expression is derived for the conditional pdf, given source location, of the signal spectral values received on an acoustic array. Examples computed for a vertical receiver array quantify localization performance degradation as an increase in the entropy of the Bayesian source-location posterior. The effects of model bias, model spectral uncertainty, and medium variability as determined by the internal wave power spectrum are quantified separately and jointly. Potential extensions to more general models of medium variability are discussed. [Work supported by ONR.]

4:20

1pSP9. Passive acoustic source localization with multiple horizontal arrays in shallow water. Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no), Peter Gerstoft, and William S. Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

This paper considers concurrent matched-field processing of data from multiple, spatially separated acoustic arrays with application to towed-source data received on two bottom-moored horizontal line arrays from the SWellEx-96 shallow water experiment. Matched-field processors are derived for multiple arrays and multiple-snapshot data using maximum-likelihood estimates for unknown complex-valued source strengths and unknown error variances. Starting from a coherent processor where phase and amplitude is known between all arrays, likelihood expressions are derived for various assumptions on relative source spectral information (amplitude and phase at different frequencies) between arrays and from snapshot to snapshot. Processing the two arrays with a coherent-array processor (with inter-array amplitude and phase known) or with an incoherent-array processor (no inter-array spectral information) both yield improvements in localization over processing the arrays individually. The best results with this data set were obtained with a processor that exploits relative amplitude information but not relative phase between arrays. The localization performance improvement is retained when the multiple-array processors are applied to short arrays that individually yield poor performance.

4:40

1pSP10. Geophysical inversion by dictionary learning. Michael J. Bianco and Peter Gerstoft (Marine Physical Lab., Univ. of California San Diego, Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92037, mbianco@ucsd.edu)

Dictionary learning, a form of unsupervised machine learning, has recently been applied to ocean sound speed profile (SSP) data to obtain compact dictionaries of shape functions which explain SSPs using as few as one non-zero coefficient. In this presentation, the results of this analysis and potential applications of dictionary learning techniques to the inversion of real acoustic data are discussed. The estimation of true geophysical parameters from acoustic observations often is an ill-conditioned problem that is regularized by enforcing prior constraints such as sparsity or energy penalities, and by reducing the size of the parameter search. Traditionally, empirical orthogonal functions (EOFs) and overcomplete wavelet and curvelet dictionaries have been used to represent complex geophysical structures with few parameters. Using the K-SVD dictionary learning algorithm, the representation of ocean SSP data is significantly compressed relative to EOF analysis. The regularization performance of these learned dictionaries is evaluated against EOFs in the estimation of ocean sound speed structure from ocean acoustic observations and the limitations of such unsupervised methods are considered.

SUNDAY AFTERNOON, 25 JUNE 2017 ROOM 306, 1:20 P.M. TO 5:00 P.M.

Session 1pUWa

Underwater Acoustics: Ambient Sound in the Ocean

Peter Gerstoft, Chair

SIO Marine Phys. Lab. MC0238, Univ. of California San Diego, 9500 Gillman Drive, La Jolla, CA 92093-0238

Contributed Papers

1:20

1pUWa1. Arctic soundscape measured with a drifting vertical line array. Emma Reeves, Peter Gerstoft, Peter F. Worcester, and Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, ecreeves@ucsd.edu)

The soundscape in the eastern Arctic was studied from April to September 2013 using a 22 element vertical hydrophone array as it drifted from near the North Pole (89° 23'N, 62° 35'W) to north of Fram Strait (83° 45'N 4° 28' W). The hydrophones recorded for 108 minutes on six days per week with a sampling rate of 1953.125 Hz. After removal of data corrupted by nonacoustic flow-related noise, 19 days throughout the transit period were analyzed. Major contributors include broadband and tonal ice noises, seismic airgun surveys, and earthquake *T* phase arrivals. Statistical spectral analyses show a broad peak in power at about 15 Hz similar to that previously observed and a mid-frequency fall off of about 20 dB/decade. Observations of the median noise levels with depth demonstrate the change in dominant noise sources between high (200-500 Hz) and low (10-50 Hz) frequencies as the array transited southward. The median noise levels observed are among the lowest of the sparse observations in the eastern Arctic, but comparable to noise levels reported in the western Arctic.

1:40

1pUWa2. Acoustic measurements of a controlled gas seep. Kevin M. Rychert and Thomas C. Weber (Ocean Eng., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, krychert@ccom.unh.edu)

To verify existing models for conversion of acoustic target strength to estimates for the total volume of methane gas released from the seafloor through the water column, a synthetic seep system was designed and fabricated. This system creates individual bubbles of a specific sizes most commonly found in gaseous methane seeps, 1 to 5 mm radii, which can be released at any interval and at any water depth. The synthetic seep system was deployed off the coast of New Hampshire in an approximate depth of 50 m. Acoustic backscatter from 10 to 100 kHz was collected by steaming over the synthetic seep multiple times, each with a predetermined and calibrated bubble size created by the system at depth. These data represent a direct field measurement which tests models describing bubble size evolution during ascent through the water column, as well as models for acoustic scattering from bubbles of different sizes. Validating these models directly tests the ability of broadband sonar systems to acoustically monitor the transport of gas from the seabed to the atmosphere.

2:00

1pUWa3. Environmental indicators from underwater soundscapes and simultaneously collected non-acoustic data. Simon E. Freeman (Underwater Acoust. and Signal Processing, US Naval Res. Lab., 7038 Old Brentford Rd., Alexandria, VA 22310, simon.freeman@gmail.com) and Lauren A. Freeman (Remote Sensing Div., US Naval Res. Lab., Washington, DC)

Drawing environmental conclusions from underwater acoustic recordings alone can to be challenging as most received sounds, especially in shallow water, are from sources that cannot easily be classified. This presentation will discuss two ongoing projects in which soundscapes were recorded simultaneously with non-acoustic, validating environmental data. In one case, acoustic data were collected simultaneously with bottom composition, organism censuses, and night-time time-lapse imagery over shallow reefs in the Hawaiian Islands. Multivariate analysis showed that areas defined by a "cool tropics" oceanographic regime grouped along a principal component parallel to monotonically increasing acoustic frequency: protected or more remote sites produced soundscapes that featured greater levels of low frequency (<2 kHz) biological sound. Degraded and/or algae dominated sites produced soundscapes featuring higher levels of high frequency (2-20 kHz) sound. The second case involves experimental work using hyperspectral optical data and simultaneously obtained acoustic recordings in shallow water, combined with in-situ spectrophotometer readings and visual benthic surveys. Optical reflectance and sound are produced through completely different mechanisms, yet the underlying environmental phenomena we wish to evaluate exhibit both. This presentation will discuss the potential of fusing these data in order to enhance our understanding of the shallow water environment. [Work supported by ONR.]

2:20

1pUWa4. A wind-driven noise model in deep water. Fenghua Li, Dong Xu, and Yonggang Guo (State Key Lab. of Acoust., Inst. of Acoust., CAS, No. 21 Beisihuanxi Rd., Beijing 100190, China, lfh@mail.ioa.ac.cn)

Surface processes, including non-linear surface wave interactions and bubble oscillations, dominate the deep water ambient noise in the absence of shipping and wildlife. A single-parameter noise model, determined by the wind speed above the ocean surface, is proposed to interpret the deep ocean acoustics from 1 Hz to 4 kHz. Below hundreds of hertz, the sound is primarily generated by surface wave interactions, which is the function of surface wave spectrum. A theoretical expression of wave spectrum is proposed to contain the gravity and capillary waves, which is in accord with the features of measured wave spectra. The noise spectrum for frequency above hundreds of hertz due to the effective bubble oscillation within the bubble cloud is proposed. The noise spectrum has been estimated as a function of frequency and wind speed based on available information of the bubble distribution. The model/data comparison shows that the proposed single-parameter noise model are in reasonable agreement with the data. [Work supported by National Natural Science Foundation of China, Grant No. 11125420.]

2:40

1pUWa5. Simulations of the influence of sound speed profile and sensor configuration in the measurement of radiated noise from ships in deep and shallow waters. Christian Audoly (CEMIS/AC, DCNS Res., Le Mourillon, BP 403, Toulon 83055, France, christian.audoly@wanadoo.fr)

The ISO TC43/ SC3 standardization committee for underwater acoustics was launched a few years ago, one of the priorities being the availability of internationally agreed procedures to measure radiated noise from ships. A first step was achieved by the publication of a first standard applicable for deep waters. However, in order to study the impact of shipping noise on marine fauna in wide maritime areas, it is necessary to input sound source levels in the form of equivalent monopoles, instead of a "radiated noise level," which is affected by the interaction with sea surface and sea floor. Therefore, the committee is currently working on correction terms to remove these effects. In that context, the objective of the present study is to determine the influence of sound speed profile and sensor configuration on the sound source level estimation, using numerical simulator. The first case study is the correction term in deep waters: here, we look at the influence of sound speed profile, whereas in previous studies, a constant speed of sound is generally assumed. The second case study deals with shallow waters: here, the main purpose is to compare different sensor configuration (number and distribution in the water column).

3:00

1pUWa6. Underwater noise footprint measurements on a survey vessel. Alex Brooker (Clarke Saunders Assoc. Acoust., Winchester, Hampshire, United Kingdom) and Victor F. Humphrey (ISVR, Univ. Southampton, Highfield, Southampton SO17 1BJ, United Kingdom, vh@isvr.soton.ac.uk)

The potential impact of man-made underwater noise on the marine environment is receiving increased attention. Shipping is one of the main sources of such anthropogenic noise. In order to understand the underwater soundscape considerable effort is being placed on generating underwater noise maps, based on using AIS data to provide details of vessel locations and operational characteristics. A key input for noise mapping models is an adequate knowledge of the source strength and characteristics for each vessel. Currently the sources are usually assumed omnidirectional, given the limited data on the true vessel radiation pattern. As part of the EU SONIC (Suppression of underwater Noise Induced by Cavitation) project measurements were undertaken on a small survey vessel, operating under realistic conditions at sea in shallow water. An autonomous recorder was used to measure the sound pressure as a function of range and azimuth. The vessel made a repeated runs past the autonomous recorder for a variety of different ranges. This has enabled the vessel noise footprint to be measured as a function of frequency and speed for the vessel, showing how the azimuthal characteristics change with frequency.

3:20–3:40 Break

3:40

1pUWa7. Developing an essential ocean variable for the acoustic environment. Peter L. Tyack (Biology, Univ. of St. Andrews, Sea Mammal Res. Unit, Scottish Oceans Inst., St. Andrews, Fife KY16 8LB, United Kingdom, plt@st-andrews.ac.uk) and A Partnership for Observation of the Global Oceans International Quiet Ocean Experiment Working Group (Plymouth Marine Lab., Plymouth, United Kingdom)

The ocean science community has invested heavily in coordinating systems developed to make ocean observations. The Global Ocean Observing System (GOOS) is a research program developed to coordinate ocean observing systems. Expert panels identify requirements for the systems in terms of Essential Ocean Variables (EOVs). The absence of sound in the list of EOVs should be striking to most ocean acousticians. Sound propagates so well in the ocean that it is the best way to probe the marine environment over long distances. Marine organisms have evolved ways to use the physics of underwater sound for biosonar, to communicate, and to orient. During the industrial age, humans have developed similar tools, and sources such as commercial shipping have elevated ocean noise. A working group of the Partnership for Observation of the Global Oceans and linked to the International Quiet Ocean Experiment has developed a description of an Essential Ocean Variable for the Acoustic Environment designed to facilitate monitoring of the sound field of the oceans globally over decades, to model how human and natural sources create the sound field, and to define effects of changes in sound fields on marine life at the individual, population, and ecosystem levels.

4:00

1pUWa8. Characteristics of snapping shrimp noise in the northeastern East China Sea. Zhuqing Yuan (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093, z9yuan@ucsd.edu), Chomgun Cho (Scripps Inst. of Oceanogr., San Diego, CA), Hee-Chun Song, and William S. Hodgkiss (Scripps Inst. of Oceanogr., La Jolla, CA)

Snapping shrimp sounds are a dominant source of high-frequency ambient noise (e.g., > 1 kHz) in temperate and tropical coastal waters of depths less than 60 m. Surprisingly, a recent shallow water experiment conducted in the northeastern East China Sea (SAVEX15) revealed an abundance of snapping shrimp sounds in approximately 100-m deep shallow water from two 16-element vertical line arrays (VLAs) deployed over 10 days. Our preliminary analysis indicates the pressure amplitude statistics fits a SαS (symmetric alpha-stable) distribution due to a heavy tail over the commonly assumed Gaussian distribution, while the temporal statistics of shrimp snaps detected above some threshold appears to fit a non-homogeneous Poisson process. In addition, the VLAs allow for localization of individual snapping shrimp. In this paper, the temporal and spatial variability of the noise characteristics dominated by snapping shrimp sounds are investigated in the northeastern East China Sea.

4:20

1pUWa9. Tsunami excitation of the Ross Ice Shelf, Antarctica. Peter Gerstoft (SIO, UCSD, 9500 Gillman Dr., La Jolla, CA 92093-0238, gerstoft@ucsd.edu), Peter Bromirski (SIO, UCSD, San Diego, CA), Zhao Chen (SIO, UCSD, La Jolla, CA), Ralph A. Stephen (WHOI, Woods Hole, MA), Rick C. Aster (Dept. of GeoSci., Colorado State Univ., Fort Collins, CO), Doug A. Wiens (Dept. of Earth and Planetary Sci., Washington Univ. in St. Louis, St. Louis, MO), and A. Nyblade (Dept. of GeoSci., Pennsylvania State Univ., State College, PA)

The responses of the Ross Ice Shelf (RIS) to the September 16, 2015 8.3 Mw Chilean earthquake tsunami (>75 s period) and infragravity (IG) waves (50-300 s period) were recorded by a 34 element broadband seismic array deployed on the RIS for one year from November 2014. Tsunami and IG-generated signals travel from the RIS front as water-ice coupled flexural waves at gravity wave speeds (~70 m/s). Displacements across the RIS are affected by gravity wave incident direction, bathymetry under and north of RIS, and water and ice shelf thickness/properties. Horizontal displacements are about 5 times larger than vertical, producing extensional motions that may facilitate expansion of existing fractures. Excitation is continuously observed throughout the year, with horizontal displacements highest during the austral winter (>20 cm). Because flexural waves exhibit weak attenuation, significant flexural wave energy reaches the grounding zone. Flexural waves provide year-round excitation of the RIS that likely promotes iceberg calving and thus ice shelf evolution. Understanding the ocean-ice shelf mechanical interactions is important to reduce the uncertainty in the global sea level rise.

4:40

1pUWa10. Modeling on low-frequency underwater noise radiated from a typical fishing boat based on measurement in shallow water. Zilong Peng, Bin Wang, Jun Fan, and Kaiqi Zhao (Shanghai Jiao Tong Univ., 800 Dongchuan Rd., Minhang District, Shanghai, Shanghai 200240, China, zlp_just@sina.com)

As evidenced in documents during the past decades, the impact of man-made underwater noise on the marine environment has always attracted more and more interest of the global researchers. About ten kinds of models have been proposed to predict underwater radiated noise (URN) by ships, and most of which are applicable above 100 Hz. This paper is aimed to modeling the low-frequency URN. Extensive measurements were made of the URN of a small fishing boat (length 43 m, displacement 500 tons) at South China Sea. The URN data show the high-level noise below 100 Hz is mainly contributed by the mechanical noise (e.g., main engine and service diesel generator) and propeller cavitation, and performs complex varying characteristics with the speed. The effect on the Transmission Loss (TL) from the sound-speed profile and bottom has been analyzed, compared with the empirical function, which show the estimated TL has an important influence on the spectral source levels (SSLs). Inspired by the method in AQUO (Achieve QUieter Oceans) project, a predicted model applied to typical fishing boat was built.

SUNDAY AFTERNOON, 25 JUNE 2017 ROOM 309, 1:20 P.M. TO 5:40 P.M.

Session 1pUWb

Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, Structural Acoustics and Vibration, Physical Acoustics, and Biomedical Acoustics: Passive Sensing, Monitoring, and Imaging in Wave Physics II

Karim G. Sabra, Cochair
Mechanical Engineering, Georgia Institute of Technology, 771 Ferst Drive, NW, Atlanta, GA 30332-0405

Philippe Roux, Cochair
ISTerre, University of Grenoble, CNRS, 1381 rue de la Piscine, Grenoble 38041, France

Invited Papers

1:20

1pUWb1. Estimation of the entropy of seismic ambient noise: Application to passive imaging. Leonard Seydoux, Nikolai Shapiro (Institut de Physique du Globe de Paris, UMR CNRS 7154, Paris, France), and Julien de Rosny (ESPCI Paris, PSL Res. Univ., CNRS, Institut Langevin, 1 rue Jussieu, Paris 75005, France, julien.derosny@espci.fr)

Recovering Green's functions from diffuse ambient noise correlation is an efficient technique for passive seismic imaging. However, the field diffuseness is not completely fulfilled in practice because the noise is generated in preferential areas and contaminated by some highly coherent signals due to earthquakes, for instance. Here we show that the noise entropy is a robust estimator of the diffuseness. In the first part, we directly use the entropy as a metric of seismic noise activity to detect tremors around Piton de la Fournaise Volcano or

highly coherent noise sources at continental scale using data collected by USArray seismic array. Then in the second part, we take the benefit of the entropy to propose an original equalization process based on the analysis of the covariance matrix to mitigate the effect a poorly diffuse noise field. The efficiency of the method is validated with several numerical tests. We apply the method to the data collected by the USArray, when a strong earthquake occurred. The method shows a clear improvement compared with the classical equalization to attenuate the highly energetic and coherent waves incoming from the earthquake, and allows to perform reliable travel-time measurement.

1:40

1pUWb2. Environmental seismology: What can we learn from ambient noise ? Eric Larose (ISTerre, CNRS & Univ. Grenoble-Alpes, CS 40700, GRENOBLE Cedex 9 38058, France, eric.larose@univ-grenoble-alpes.fr)

Environmental seismology consists in studying the coupling between the solid Earth and the cryosphere, or the hydrosphere, the anthropospher. In practice, we monitor the modifications of the wave propagation due to environmental forcing such as temperature and hydrology, using ambient seismic noise that constitute a continuous, cheap and relatively reproducible source of vibrations. Recent developments in data processing [1], together with increasing computational power and sensor concentration have led to original observations that allow for this new field of seismology. In this paper, we will review how we can track and interpret tiny changes in the subsurface of the Earth related to external changes from modifications of the seismic wave propagation, with application to geomechanics, hydrology, and natural hazard [2]. We will demonstrate that, using ambient noise, we can track: thermal variations in the subsoil, in buildings or in rock columns with application to damage estimation; the temporal and spatial evolution of a water table; the evolution of the rigidity of the soil constituting a landslide, and especially the drop of rigidity preceding a failure event. [1] Shapiro, N. M., Campillo, M., 2004. Geophys. Res. Lett. **31**, L7 614. [2] E. Larose *et al.*: J. Appl. Geophys. **116**, 62-74 (2015).

2:00

1pUWb3. Extracting changes in wave velocity from chaotic data. Roel Snieder (Colorado School of Mines, 1500 Illinois Str., Golden, CO 80401, rsnieder@mines.edu)

Using seismic interferometry it is possible to extract the Green's function from recorded measurements of noise or other incoherent signals, it is a way of creating order out off chaos. Since the noise is present at all times, this provides a way to measure seismic velocity changes over time. The seismic velocity in the subsurface, and in buildings, is not constant in time. I will present measurements taken during, and after, the 2011 Tohoku-Oki earthquake to show that deconvolution interferometry can be used to monitor the seismic velocity in the near surface, with a great temporal resolution. The recovery of the seismic velocity typically varies as log-time, which can be related to a spectrum of relaxation processes in the earth.

2:20

1pUWb4. Imaging subsurface structures using reflections retrieved from seismic interferometry with sources of opportunity. Deyan Draganov, Yohei Nishitsuji, Boris Boullenger, Shohei Minato, Kees Wapenaar, Jan Thorbecke (Dept. of GeoSci. and Eng., Delft Univ. of Technol., Stevinweg 1, Delft 2628CN, Netherlands, d.s.draganov@tudelft.nl), Elmer Ruigrok (GeoSci., Utrecht Univ., Delft, Netherlands), Charlotte Rowe (Geophys. Group, Los Alamos National Lab., Los Alamos, NM), Bob Paap, Arie Verdel (TNO, Utrecht, Netherlands), and Martin Gomez (CNEA, Buenos Aires, Argentina)

The reflection seismic method is the most frequently used exploration method for imaging and monitoring subsurface structures with high resolution. It has proven its qualities from the scale of regional seismology to the scale of near-surface applications that look just a few meters below the surface. The reflection method uses controlled active sources at known positions to give rise to reflections recorded at known receiver positions. The reflections' two-wave travel time is used to extract desired information about and image the subsurface structures. When active sources are unavailable or undesired, one can retrieve body-wave reflections from application of seismic interferometry (SI) to sources of opportunity—quakes, tremors, ambient noise, or even man-made sources not connected to the exploration campaign. We show examples of imaging of subsurface structures using reflections retrieved from quakes and ambient noise. We apply SI by autocorrelation to global earthquake to image seismic and aseismic parts of the Nazca plate and the Moho at these places, SI by multidimensional deconvolution to P-wave coda from local earthquakes to image the Moho and the crust at the same places, and SI by autocorrelation to deep moonquakes to image the lunar Moho and to ambient noise to monitor CO_2 sequestration.

2:40

1pUWb5. Passive elastography: A shear wave tomography of the human body. Stefan Catheline (INSERM U1032, 151 cours albert thomas, Lyon 69003, France, stefan.catheline@inserm.fr)

Elastography, sometimes referred as seismology of the human body, is an imaging modality recently implemented on medical ultrasound systems. It allows to measure shear waves within soft tissues and gives a tomography reconstruction of the shear elasticity. This elasticity map is useful for early cancer detection. A general overview of this field is given in the first part of the presentation as well as latest developments. The second part, is devoted to the application of time reversal or noise correlation technique in the field of elastography. The idea, as in seismology, is to take advantage of shear waves naturally present in the human body due to muscles activities to construct shear elasticity map of soft tissues. It is thus a passive elastography approach since no shear wave sources are used.

3:00

1pUWb6. Relocating drifting sensor networks with ambient noise correlations. Brendan Nichols, James S. Martin (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr. NW, Atlanta, GA 30309, bnichols8@gatech.edu), Christopher M. Verlinden (Phys., U.S. Coast Guard Acad., La Jolla, California), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

A network of drifting sensors, such as hydrophones mounted to freely drifting buoys, can be used as an array for locating acoustic sources underwater. However, for accurate localization of such a source using coherent processing, the positions of the sensors need to be known to a high degree of accuracy, typically more accurately than provided by dead reckoning or GPS alone. Past work has demonstrated the inter-sensor distances can be obtained from long-term ambient noise correlations on fixed arrays [Sabra *et al.*, IEEE J. Ocean Engineering, 2005, 30]. Here, the approach was extended for tracking drifting sensor motion by combining a stochastic search algorithm with ambient noise correlation processing. Optimization of the stochastic search method was explored and performance compared to acoustic data collected from a volumetric hydrophone vs. vector sensor array deployed in the Long Island Sound.

3:20–3:40 Break

3:40

1pUWb7. Passive sensing of head wave propagation in the ambient noise field and its implications for geoacoustic inversion. John Gebbie (Metron, Inc., 1900 SW 4th Ave., Ste. 89-01, Portland, OR 97201, gebbie@metsci.com) and Martin Siderius (Portland State Univ., Portland, OR)

Under certain conditions, the ambient noise field can produce and amplify head waves with unique propagation characteristics; these are detectable with a vertical line array and can be analyzed to extract geoacoustic information. Head wave phenomenon in the seabed can be observed using point sources, but these arrivals are usually very weak and difficult to detect ahead of the strong direct-path arrival. In contrast, surface-generated noise from wind and breaking waves are effectively a planar source that insonify the entire seabed at every angle and from every direction. Ambient noise head waves are conical waves that are generated when noise first reaches the seabed at the critical angle, and are amplified upon each subsequent interaction with the seabed. This interaction involves splitting the incident wave into water-borne and seabed-borne components, each separated in time by a fixed lag. A vertical line array can observe this phenomenon by cross-correlating beams steered upward and downward at the critical angle of the seabed. Together, the steering angle and time lag depend on the seabed critical angle and ray cycle time through the waveguide. Experimental results will be presented along with full-wave simulations that help illustrate the phenomena.

4:00

1pUWb8. A tomography experiment using ships as sources of opportunity. William A. Kuperman, Bruce Cornuelle, Kay L. Gemba, William S. Hodgkiss, Jit Sarkar, Jeffery D. Tippmann, Christopher M. Verlinden (Scripps Inst. of Oceanogr., UCSD, Marine Physical Lab., La Jolla, CA 92093-0238, wkuperman@ucsd.edu), and Karim G. Sabra (College of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

An experiment was performed in the Santa Barbara Channel using four vertical acoustic receive arrays placed between the sea lanes of in- and outgoing shipping traffic. The purpose of the experiment was to determine whether these sources of opportunity can be utilized for tomographic inversion of water column properties. The environment was continuously monitored throughout the duration of the experiment. Ship tracks were obtained from the Automatic Identification System (AIS). Processing was developed to extract relative time delays between the arrays from the ships' random radiation fields. This information, together with AIS constraints were used for inversion. Initial results are presented that also include an error analysis of the inversion.

4:20

1pUWb9. Deducing environmental properties from broadband matched-mode processing ambiguity surface striations generated from baleen whale data. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Julien Bonnel (ENSTA, Brest cedex 9, France), and Catherine L. Berchok (Marine Mammal Lab., Alaska Fisheries Sci. Ctr., Seattle, WA)

Numerous multi-year shallow-water recordings of several species of baleen whale have been obtained from shallow-water arctic environments, including the Bering Sea. When non-linear time sampling is applied to the single-hydrophone data, these broadband signals yield individual normal mode arrivals, which in turn permit incoherent matched-mode processing (MMP) techniques to be applied for source localization and geoacoustic inversion. When continuously broadband MMP ambiguity surfaces are constructed from pairs of modes and plotted as a function of range and frequency, both the mainlobe and sidelobes form striations that embed information about the type and amount of environmental mismatch present between the modeled and true environment. These striations are useful for identifying bandwidths of inversion-quality data within whale calls. Acoustic invariant theory explains how mismatched waveguide replicas from simple environmental models, when applied to sufficiently low-frequency data, produce ambiguity surface striations that reveal the true bottom interface sound speed and bottom sound speed gradient. During summer, whenever highly downward-refracting sound speed profiles exist, mismatched MMP striations also contain information about the sound speed gradient. Examples of this visual approach to inversion are shown on endangered North Pacific Right whale "gunshot" signals. [Work sponsored by the North Pacific Research Board.]

4:40

1pUWb10. Environmental characterization in the Chukchi Sea using Bayesian inversion of bowhead whale calls. Graham A. Warner (JASCO Appl. Sci., 2305-4464 Markham St., Victoria, BC V8Z 7X8, Canada, graham.warner@jasco.com), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and David E. Hannay (JASCO Appl. Sci., Victoria, BC, Canada)

This paper estimates seabed and water-column properties of a shallow-water site in the Chukchi Sea using bowhead whale calls recorded on asynchronous ocean-bottom hydrophones. Up- and down-swept bowhead whale calls were recorded on a cluster of seven hydrophones within a 5 km radius. The calls excited multiple propagating modes, with modal dispersion controlled by environmental properties and whale-recorder range. Frequency-dependent mode arrival times for nine whale calls are inverted using a trans-dimensional (trans-D) Bayesian approach that estimates the whale locations (easting and northing) and range-independent environmental properties (sound-speed profile, water depth, and seabed geoacoustic profiles). The trans-D inversion allows the data to determine the most appropriate environmental model parameterization in terms of the number of sound-speed profile nodes and subbottom layers. The inversion also estimates each whale-call instantaneous frequency function, relative recorder clock offsets, and residual-error standard deviation, and provides uncertainty estimates for all model parameters and parameterizations. The sound-speed profile is found to be poorly resolved, but water depth and upper sediment-layer thickness and sound speed are reasonably well resolved. Model estimates and uncertainties are compared to those from separate inversions involving airgun dispersion and vessel noise data collected nearby (which also represent sources of opportunity).

5:00

1pUWb11. Monitoring bubble production in a seagrass meadow using a source of opportunity. Paulo Felisberto (LARSyS, Univ. of Algarve, Faro, Faro, Portugal), Orlando C. Rodríguez, João P. Silva, Sergio Jesus (LARSyS, Univ. of Algarve, Campus de Gambelas - Universidade do Algarve, Faro, N/A PT-8005-139, Portugal, orodrig@ualg.pt), Hugo Quental-Ferreira, Pedro Pousão-Ferreira, Maria Emília Cunha (IPMA - Instituto Português do Mar e da Atmosfera, EPPO, Olhão, Portugal), Carmen B. de los Santos, Irene Olivé, and Rui Santos (Marine Plant Ecology Res. group, Ctr. of Marine Sci. of Univ. of Algarve, Faro, Portugal)

Under high irradiance, the photosynthetic activity of dense seagrass meadows saturates the water forming oxygen bubbles. The diel cycle of bubble production peaks at mid-day, following light intensity pattern. It is well known that bubbles strongly affect the acoustic propagation, increasing signal attenuation and decreasing the effective water sound speed, noticeable at low frequencies. Thus, the diurnal variability of bubbles may show an interference pattern in the spectrograms of low frequency acoustic signals. In an experiment conducted in July 2016 at the Aquaculture Research Station of the Portuguese Institute for the Sea and Atmosphere in Olhão, Portugal, the spectrograms of low frequency (<20 kHz) broadband noise produced by water pumps in a pond of 0.48 ha covered by the seagrass Cymodocea nodosa showed interference patterns that can be ascribed to the variability of the sound speed in the water. Preliminary analysis suggests that the daily cycle of bubble concentration can be inferred from these interference patterns.

Contributed Paper

5:20

1pUWb12. Ambient noise correlations on a mobile, deformable array. Perry Naughton (Elec. and Comput. Eng., Univ. of California, San Diego, 9500 Gilman Dr., San Diego, CA 92103, pnaughto@eng.ucsd.edu), Philippe Roux (ISTerre, Université de Grenoble Alpes, Grenoble, France), Riley Yeakle (Elec. and Comput. Eng., Univ. of California, San Diego, San Diego, CA), Curt Schurgers (Qualcomm Inst., Calit2, Univ. of California, San Diego, San Diego, CA), Ryan Kastner (Comput. Sci. and Eng., Univ. of California, San Diego, San Diego, CA), Jules Jaffe, and Paul Roberts (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA)

This presentation describes a demonstration of ambient acoustic noise processing on a set of free floating oceanic receivers whose relative positions vary with time. We show that we are able to retrieve information that is relevant to the travel time between the receivers. With thousands of short time cross-correlations of varying distance, we show that on average, the decrease in amplitude of the noise correlation function with increased separation follows a power law. This suggests that there may be amplitude information that is embedded in the noise correlation function. We develop an incoherent beamformer, which shows that it is possible to determine a source direction using an array with moving elements and large element separation. We show how the noise correlation function varies in the presence of a boat with a known GPS trajectory, and how this information can be used recover the relative geometry of the deformable array. This work indicates that the relative geometry of an array can be estimated using only passive signals and the automatic identification system already present in many coastal communities.

Session 1pUWc

Underwater Acoustics: Topics in Underwater Acoustics (Poster Session)

Vaibhav Chavali, Chair
Electrical Engineering, George Mason University, 4217 University Dr., Fairfax, VA 22030

All posters will be on display from 1:20 p.m. to 4:20 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 1:20 p.m. to 2:50 p.m. and authors of even-numbered papers will be at their posters from 2:50 p.m. to 4:20 p.m.

Contributed Papers

1pUWc1. Determining the bottom surface in the randomly inhomogeneous media. Andrei Sushchenko (School of Natural Sci., Far Eastern Federal Univ., Sukhanova 8, Vladivostok, Primorskii krai 690090, Russian Federation, sushchenko.aa@dvfu.ru), Igor Prokhorov (Inst. of Appl. Mathematics FEB RAS, Vladivostok, Russian Federation), and Kristina Sushchenko (School of Natural Sci., Far Eastern Federal Univ., Vladivostok, Russian Federation)

The authors study a problem of determining the bottom topography of a fluctuating ocean using the data of side-scan sonars. Based on a kinetic model of acoustic radiative transfer authors obtain a formula for determining a function describing small deviations of the bottom surface from a middle level. The impulse parcels of source and width of the directivity pattern of the receiving antenna are constructed unfocused sebottom image. For solving it, authors used iterative algorithm for focusing objects on the seabottom. Numerical experiments have been done on modeling data that demonstrate the accuracy of the obtained formula. Numerical analysis of the volume scattering influence is done. The volume scattering filter allows to reconstruct sea bottom relief from long range, e.g., signal from 150 m includes more than 50% of volume scattering, hence object recognizing is not possible without filtering. The width of directivity pattern affects to the object defocussing. This effect is increased with slant range increasing. Moreover, authors designed the algorithm of determining shaded areas on the sea bottom. It allows to recognize each invisible point on the sea bottom in case of non-static source. Thus, authors researched influence of volume scattering on the seabottom relief reconstruction.

1pUWc2. Interferometric reconstruction of plate waves from cross correlation of diffuse field on a thin aluminum plate. Aida Hejazi Nooghabi (Univ. of Pierre and Marie Curie, 4, Pl. Jussieu Case 129, T.46-00, Et.2, Paris 75252, France, aida.hejazi@gmail.com), Julien de Rosny (Institut Langevin, Paris, France), Lapo Boschi (Univ. of Pierre and Marie Curie, Paris, France), and Philippe Roux (Laboratoire ISTERRE, Grenoble, France)

This study contributes to evaluating the robustness and accuracy of Green's function (GF) reconstruction by cross-correlation of noise, disentangling the respective roles of ballistic and reverberated ("coda") signals. We conduct a suite of experiments on a highly reverberating thin aluminum plate, where we generate an approximately diffuse flexural wavefield. We validate ambient-noise theory by comparing cross correlation to the directly measured Green's function. We develop analytically a theoretical model, predicting the dependence of the symmetry of the cross correlations on the number of sources and signal-to-noise ratio. We validate this model against experimental results. We next study the effects of cross-correlating our data over time windows of variable length, possibly very short, and taken at different points in the coda of recordings. We find that, even so, a relatively dense/uniform source distribution could result in a good estimate of the GF; we demonstrate that this window does not have to include the direct-arrival signal for the estimated GF to be a good approximation of the exact one. Afterwards, we explicitly study the role of non-deterministic noise on cross correlations and establish a model which confirms that the relative effect of noise is stronger when the late coda is cross-correlated.

1pUWc3. Reflecting boundary conditions for interferometry by multidimensional deconvolution. Cornelis Weemstra, Kees Wapenaar (Dept. of GeoSci. and Eng., Delft Univ. of Technol., Stevinweg 1, Delft 2628 CN, Netherlands, kweemstra@gmail.com), and Karel N. van Dalen (Dept. of Structural Eng., Delft Univ. of Technol., Delft, Netherlands)

Seismic interferometry (SI) takes advantage of existing (ambient) wavefield recordings by turning receivers into so-called "virtual-sources." The medium's response to these virtual sources can be harnessed to image that medium. Applications of SI include surface-wave imaging of the Earth's shallow subsurface and medical imaging. Most interferometric applications, however, suffer from the fact that the retrieved virtual-source responses deviate from the true medium responses. The accrued artifacts are often predominantly due to a non-isotropic illumination of the medium of interest, and prohibit accurate interferometric imaging. Recently, it has been shown that illumination-related artifacts can be removed by means of a so-called multidimensional deconvolution (MDD) process. However, The current MDD formulation, and hence method, relies on separation of waves traveling inward and outward through the boundary of the medium of interest. As a consequence, it is predominantly useful when receivers are illuminated from one side only. This puts constraints on the applicability of the current MDD formulation to omnidirectional wavefields. We present a modification of the formulation of the theory underlying SI by MDD. This modification eliminates the requirement to separate inward-and outward propagating wavefields and, consequently, holds promise for the application of MDD to non-isotropic, omnidirectional wavefields.

1pUWc4. Status and results from cabled hydrophones arrays deployed in deep sea off East Sicily (EMSO-ERIC node). Giorgio Riccobene, Francesco Caruso (Laboratori nazionali del Sud, Istituto Nazionale di Fisica Nulceare, Catania, Italy), Salvatore Viola (Laboratori nazionali del Sud, Istituto Nazionale di Fisica Nulceare, Via S. Sofia 62, Catanoa, Italy, sviola@lns.infn.it), Francesco Simeone (Sez. Roma 1, Istituto Nazionale di Fisica Nulceare, Roma, Italy), Sara Pulvirenti, Virginia Sciacca (Laboratori nazionali del Sud, Istituto Nazionale di Fisica Nulceare, Catania, Italy), Carmelo Pellegrino (Sez. Bologna, Istituto Nazionale di Fisica Nulceare, Bologna, Italy), Fabrizio Speziale (Laboratori nazionali del Sud, Istituto Nazionale di Fisica Nulceare, Catania, Italy), Fabrizio Ameli (Sez. Roma 1, Istituto Nazionale di Fisica Nulceare, Roma, Italy), Giuseppa Buscaino, Salvatore Mazzola (CNR-IAMC, Capo Granitola (TP), Italy), Francesco Filiciotto (CNR-IAMC, Messina, Italy), Rosario Grammauta (CNR-IAMC, Capo Granitola (TP), Italy), Gaetano Licitra (ARPAT, Pisa, Italy), Giorgio Bellia (Laboratori nazionali del Sud, Istituto Nazionale di Fisica Nulceare, Catania, Italy), Gianni Pavan (Univ. of Pavia, Pavia, Italy), Davide Embriaco (INGV, La Spezia, Italy), Paolo Favali, Laura Beranzoli, Giuditta Marinaro, Gabriele Giovanetti (INGV, Roma, Italy), Francesco Chierici (IRA, INAF, Bologna, Italy), Giuseppina Larosa (Laboratori nazionali del Sud, Istituto Nazionale di Fisica Nulceare, Catania, Italy), Antonio D'Amico (NIKHEF, Amsterdam, Netherlands), and Elena Papale (CNR-IAMC, Capo Granitola (TP), Italy)

Since 2005 a cabled deep-sea infrastructure is operative at 2100 m water depth, 25 km off the port of Catania (Sicily). The infrastructure, under continuous improvement, is the first operative cabled node of the EMSO-ERIC, hosting several multidisciplinary observatories built in collaboration by INFN, INGV, CNR, CIBRA, and other scientific partners. Hydrophones antennas, sensitive in the range of frequencies between 1 Hz and 90 kHz, have been installed on seafloor observatories. Acoustic data are continuously digitized in situ at very high resolution, time-stamped with absolute GPS time and sent to shore in real time, through optical fiber link. Together with biological sounds, noise pollution study and monitoring were the main goals of the research. Results of multi-year monitoring of anthropogenic noise are discussed. Focus of the analysis is the noise level in the octave bands centered at 63 Hz and 125 Hz, in compliance with the EU Marine Strategy Framework Directive. The contribution of ship noise was modeled, based on their data recorded via proprietary AIS antennas, and compared to data. Noise at higher frequencies was also investigated. Detection of airguns emissions and recorded noise levels is reported. Status and coming activities at the infrastructure is also presented.

1pUWc5. Spatial distribution of sound field scattered from the rough seafloor interface. Linhui peng (Ocean Technol., Ocean Univ. of China, 238 Songling Rd., Information College, Qingdao, Shangdong 266100, China, penglh@ouc.edu.cn), Gaokun Yu, and JIanhui Lu (Ocean Technol., Ocean Univ. of China, Qingdao, Shandong, China)

The scattering coefficient of rough seafloor interface is calculated using first order perturbation theory. The interface roughness considered here is described by the isotropic power law spectrum for isotropic rough seafloor interface and anisotropic Gaussian spectrum for the rippled seafloor interface. The characteristic of spatial distribution is shown by the scattering coefficient of the scattered field with forward scattering and backward scattering, which depend on the frequency and the grazing angle of the incident wave, and the roughness of the interface. The dependence of spatial distribution of the scattered field on these parameters is analyzed by Bragg scattering of the sinusoidal interface.

1pUWc6. Computation of acoustic wave responses due to moving underwater acoustic sources in complex underwater environments using a spectral element method. Stephen Lloyd, Chanseok Jeong (Civil Eng., The Catholic Univ. of America, 620 Michigan Ave., N.E., Washington, DC 20064, 34lloyd@cua.edu), Hom Nath Gharti, and Jeroen Tromp (GeoSci., Princeton Univ., Princeton, NJ)

This work presents a new numerical approach for computing underwater acoustic wave responses due to moving underwater acoustic sources in complex underwater environments using a Spectral Element Method (SEM). The SEM is similar to the Finite Element Method (FEM), but uses a higher-order shape function with Gauss-Lobatto-Legendre quadrature, naturally creating a diagonal mass matrix. Thus, we can use fast explicit time integration, taking advantage of a diagonal mass matrix, without compromising accuracy. Therefore, the SEM is much more suitable for large-scale parallel 3D time domain wave analyses than the conventional FEM. In our numerical experiments, we used a large-scale parallel SEM wave simulator, SPECFEM3D. We verified the SEM solution of acoustic (fluid pressure) waves in a 3D acoustic fluid setting of an infinite extent, induced by a moving point source, by using its analytical counterpart. Numerical experiments showed that our tool accurately accommodates wave behavior at fluid-solid interfaces of complex geometries and infinite extents of water and solids, truncated using absorbing boundary conditions. Due to such versatility, our tool can be used for forward and inverse acoustic wave analyses in any complex underwater systems of large extents (e.g., shallow water and deep ocean).

1pUWc7. Robust multiple focusing with adaptive time-reversal mirror using a genetic algorithm. Gi Hoon Byun and Jea Soo kim (Korea Maritime and Ocean Univ., Korea Maritime Univ., Dongsam 2-dong, Yeongdo-gu, Busan 606-791, South Korea, gihoonbyun77@gmail.com)

Kim and Shin [J. Acoust. Soc. Am. 115 (2), 600-606 (2003)] suggested an extension of the single constraint adaptive time-reversal mirror (ATRM) to simultaneous multiple focusing, by considering multiple constraints. In their proposed method, the optimization is performed using the linearly constrained minimum variance (LCMV) method, a well known optimization method in the field of adaptive signal processing that allows multiple linear constraints. However, highly correlated signal vectors from the probe source locations cause prominent spatial sidelobes in multiple TR focusing. In this study, a genetic algorithm is extended to LCMV method to calculate the backpropagation vector which satisfies new constraint responses. Numerical simulations demonstrate that multiple TR focusing combined with a genetic algorithm can significantly suppress sidelobes, especially when the focal locations are close to each other.

1pUWc8. Modeling method on acoustic scattering from penetrable objects using a hybrid Kirchhoff/ray approach. Bin Wang, Kaiqi Zhao, Jun Fan (School of Naval Architecture, Ocean and Civil Eng., Shanghai Jiao Tong Univ., Dongchuan Rd. 800, Shanghai, Shanghai 200240, China, lanseyifan48@sjtu.edu.cn), and Guoyin Zheng (School of Naval Architecture, Ocean and Civil Eng., Shanghai Jiao Tong Univ., Wuhan, Hubei, China)

A method is put forward to investigate the acoustic scattering from double-sided water loaded targets which is penetrable even in high frequency. The present approach is an extension of the TriKirch method which was elaborated for non-penetrable targets [J. Acoust. Soc. Am. 140(3), 1878-1886 (2016)] and complex targets are dispersed into many triangular planar facets. Reflection coefficient of plane facet is introduced to calculate the scattering amplitude of each non-rigid triangular facets combined with TirKirch method, and the total scattering amplitude of target is obtained by superposing the amplitude of all facets' contributions coherently. Double bounce (DB) contributions to the scattering is calculating by ray-tracing method. Computations are made for a double-sided water loaded finite cylindrical shell with perforated ring ribs, and are compared with the experimental results, and both results show good agreement.

1pUWc9. Determination of eigenvalue in elastic multi-layered waveguides. Wang Wei and Bin Wang (Shanghai Jiao Tong Univ., No.800, Dongchuan Rd., Minhang District, Shanghai 200240, China, wei_wang@sjtu.edu.cn)

With the extensive use of composites in vibration and noise reduction in recent years, the properties of acoustic propagation in composite plates have come to the foreground. However, traditional root-finding methods based on contour integration and finite difference show low precision and root-missing problems on calculating the dispersion curve, especially the loss factor is excessively influential and cannot be ignored. In this paper, a multimodal approach (Pagneux *et al.*, Proc. R. Soc. A (2006) 462, 1315-1339) is applied to solve the eigenvalue problem for elastic multi-layered waveguides. The longitudinal and transverse potential functions are expanded on a group of orthogonal basis respectively and a matrix equation is then derived with the boundary conditions, which can be solved after truncation at an adequate number of modes. Dispersion curves of composite plates with different loss are presented. The numerical simulation proves that it is more effective than traditional methods

1pUWc10. Characterization of arctic ambient noise environment. Rui Chen and Henrik Schmidt (Mech. Eng., MIT, Massachusetts Inst. of Technol., 77 Massachusetts Ave., Rm. 5-223, Cambridge, MA 02139, ruic@mit.edu)

Historically, ambient noise in arctic ocean is predominately produced by diffuse thermal ice cracking events or ice ridge grinding. Isotropic, range-distributed noise sources models are typically utilized to simulate this environment. However, the presence of the Beaufort Lens and changes in the arctic climate have altered its ambient noise environment. Specifically, the new noise environment consists mostly of ice cracking events which occur at discrete ranges and bearings. As a result, these noise models may no longer be adequate. This study analyzes ambient noise data collected in the Beaufort Sea during the 2016 ICEX US Navy Exercise to characterize the new arctic ambient noise environment. Points of focus include determining whether ice cracking noises in the new environment are discrete in time or continuous as is the result from analysis of the SIMI'94 arctic ambient noise data. Statistics on the ice cracking events in the new noise environment, such as the events' amplitude distribution, are also presented with the motivation of better describing the environment so that more precise models may be created.

1pUWc11. Performance of a ray-based blind deconvolution algorithm for shipping sources in the Santa Barbara channel. Nicholas C. Durofchalk (Mech. Eng., Georgia Inst. of Technol., 101 N. College Ave, Lebanon Valley College 208 Ctr. Hall, Annville, PA 17003, ncd001@lvc.edu), Juan Yang (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

This paper investigates the performance of a ray-based blind deconvolution (RBD) algorithm for sources of opportunity, such as shipping noise, in an ocean waveguide recorded on a vertical line array (VLA). The RBD algorithm [Sabra *et al.,* JASA,2010, EL42-7] relies on estimating the unknown phase of the source through wideband beamforming along a well-resolved ray path to approximate the environment's Green's Functions (or Channel Impulse responses) between the source and the VLA elements, as well with recovering the unknown radiated source signal. The RBD algorithm is tested here for shipping sources recorded in the Santa Barbara shipping channel (water depth ~550 meters). The four VLAs, with short (~ 15 meter) and long (~56 meter) apertures, were deployed between the north and south bound shipping lanes and collected acoustic data throughout a week in mid-September, 2016. Here, we discuss (1) the performance of conventional beamforming and adaptive (MVDR) beamforming when estimating ray-arrivals, (2) the ability of the RBD algorithm to deconvolve multiple VLAs using the only the same source phase estimated from a single VLA. The performance of the RBD results will be discussed in terms of accuracy of travel times and of the estimated Green's Functions.

1pUWc12. Reflection coefficient measurement using a finite-difference-injection technique. Nele Börsing, Carly Donahue, Dirk-Jan van Manen, and Johan O. Robertsson (Inst. of Geophys., Dept. of Earth Sci., ETH Zürich, Sonneggstrasse 5, NO H32, Zürich 8092, Switzerland, nele.boersing@erdw.ethz.ch)

Experimentally determining the acoustic reflection properties of materials requires accurate knowledge of the incident and reflected wave field at the reflecting interface. Consequently, a number of methods have been proposed for measuring the reflection coefficient, but most are limited to measuring only the normal incidence reflection coefficient or assume plane wave conditions. Here, we derive the reflection coefficient from pressure measurements by using a finite-difference wave field injection technique, which is applicable for a wide range of incidence angles and does not rely on the common plane wave assumption. It requires measurements at three planes parallel to the reflector and addresses two key objectives, namely, (1) the recorded wave field is separated into its incident and reflected components without the need of time-windowing, and (2) the separated components are re-datumed to the reflecting interface. The latter step comprises a forward propagation in time of the incident wave and a time-reversed backward propagation of the reflected wave. We experimentally test the methodology on laboratory data of the reflection from the free surface recorded in water and demonstrate its applicability to accurately measure the reflection coefficient for incidence angles up to 60°.

1pUWc13. Roughness parameters imaging with a multibeam echosounder. Samuel Pinson, Yann Stéphan (SHOM, SHOM, Brest 29200, France, samuelpinson@yahoo.fr), and Charles W. Holland (Penn State Univ., State College, PA)

The aim of the study is to perform quantitative imaging of the seafloor random parameters using a multibeam echosounder. More specifically we present in this communication a focus on the interface roughnesses with controlled parameters using a 3D modeling of a layered media with rough interfaces. The modeling consists of a sum of integrals over each interface of the layered medium that implies a reasonable computation cost and the possibility to perform a high number of numerical experiments. Specular reflection and backscattering are analyzed by estimating their means and variances through imaging algorithms. [Research supported by the ONR Ocean Acoustics Program.]

1pUWc14. Interface wave studies on a laboratory test facility. Gopu R. Potty, Rendhy M. Sapiie, Chris J. Small, and James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu)

Rayleigh wave measurements were made in the Interface Wave Test Facility at the University of Rhode Island to develop techniques to estimate the shear wave properties of near surface sediment. Repeating source events at various ranges spaced equally at 0.15 m from a fixed receiver (accelerometer) created a virtual source array. The source events consisted of dropping a tennis ball thereby exciting Rayleigh waves. A monitoring accelerometer was used to record each source event at a fixed distance from the source location in order to time it. This enabled to calculate the travel time of the Rayleigh waves from the monitoring receiver to the fixed receiver. The phase velocity dispersion is calculated using high resolution frequency-wavenumber processing. The shear wave speed of the sediment layers in the "sand tank" is estimated using an inversion scheme. The shear wave speed estimates is compared to direct measurements using a calibrated bender element system at selected depths. The bender element system was placed in a test tank initially filled with mason sand and the Rayleigh wave inversion system was deployed on the surface of the sand in the tank. [Work supported by Army Research Office and ONR.]

1pUWc15. Mode coherence in random matrix theory simulations. Kathleen E. Wage (George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, kwage@gmu.edu) and Lora J. Van Uffelen (Univ. of Rhode Island, Narragansett, RI)

Random matrix theory (RMT) can be used to simulate the effect of internal waves on broadband acoustic mode time series in deep water, as described by Hegewisch and Tomsovic in several papers [Europhys. Lett. 2012; J. Acoust. Soc. Am. 2013]. Using RMT, narrowband mode propagation consists of the multiplication of a series of propagator matrices designed to model the mode coupling due to internal waves at a single frequency. In a recent paper, we varied the correlation of the mode coupling matrices with frequency and examined the effect on the time series generated via Fourier synthesis [Van Uffelen & Wage, Inst. of Acoustics Conf., 2016]. This talk focuses on another key aspect of the RMT model, i.e., mode-to-mode coherence at a single frequency. Modes decorrelate as they propagate through internal waves. Building on Creamer's work, Colosi and Morozov use transport theory to analyze mode-to-mode coherence and mode energy in deep water environments [J. Acoust. Soc. Am., 2009]. This talk compares the energy and coherence results obtained with RMT methods to those from transport theory. [Work sponsored by ONR.]

1pUWc16. Global scale underwater sound modeling and data analysis. Tiago Oliveira (Woods Hole Oceanographic Inst., Singapore, Singapore), Ying-Tsong Lin, Arthur Newhall (Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu), Stephen Nichols (Penn State Univ., State College, PA), and Dave Bradley (Penn State Univ., Woods Hole, Massachusetts)

Underwater low-frequency sound can travel great distances in the oceans, and sound triggered in the sea by the mechanical energy transfer from the Earth's crust (e.g., earthquakes or volcanoes) and by the energy transfer occurring at the water surface (e.g., wave storms or ice-quakes) can be detected at thousands of kilometers from the source. However, source characterization based on recorded sound data analysis involves significant scientific challenges and uncertainties. A variety of geological and physical oceanographic features can cause horizontal refraction, reflection, and diffraction on global scale sound propagation. In this regard, three-dimensional underwater sound models are required for accurately predicting global scale sound propagation. In this work, based on a Southern Mid-Atlantic Ridge earthquake event, we show the importance of geological and physical oceanographic features in the long range propagation of oceanic sound. A three-dimensional sound propagation model using the parabolic equation (PE) approximation and the split-step Fourier (SSF) method is used. Numerical results are compared with field data recorded by hydrophones at great distances from the source. Based on the case study, a discussion and recommendation on the global scale underwater sound modeling and data analysis are presented.

1pUWc17. Investigation of error of propagated sound due to bathymetric interpolation. Erin C. Hafla, Erick Johnson (Mech. Eng., Montana State Univ., 205 Cobleigh Hall, Bozeman, MT 59717-3900, erinhafla@gmail.com), and Jesse Roberts (Sandia National Labs., Albuquerque, NM)

Paracoustic is a parallelized acoustic-wave propagation package developed by Sandia National Laboratories to model marine hydrokinetic (MHK) devices in complex environments. It solves a linearized set of the velocity-pressure partial differential equations using the finite-difference method and allows for 3D variations in medium densities, sound speeds, and bathymetry. In-situ measurements of these quantities, or the solution resolution from hydrodynamic models, are sometimes resolved at a coarser spacing than is required to accurately predict the propagated sound levels within Paracousti. This will therefore require interpolation of these quantities onto a refined grid, introducing model errors. The size of the refined grid is determined by the maximum frequency of an MHK source profile and the underwater sound speed. A single point source in a two-layer waveguide with realistic bathymetry is used to compare the model error between three interpolation schemes: nearest-neighbor, linear, and cubic. Preliminary results indicate that there are significant differences in model error due to the particular interpolation scheme used.

1pUWc18. 3D acoustic propagation modeling of the construction of the Block Island Wind Farm. Anthony F. Ragusa, Gopu R. Potty, James H Miller (Ocean Eng., Univ. of Rhode Island, 15 Receiving Rd., Narragansett, RI 02882, afragusa@my.uri.edu), Ying-Tsong Lin, Arthur Newhall (Woods Hole Oceanographic Inst., Woods Hole, MA), Kathleen J. Vigness-Raposa (Marine Acoust., Inc., Middletown, RI), Jennifer Giard (Marine Acoust., Inc., Narragansett, RI), Michael Ross, and Jesse Roberts (Sandia National Labs., Albuquerque, NM)

The Block Island Wind Farm (BIWF) consists of five turbines in water depths of approximately 30m. The substructure for the BIWF turbines consists of jacket type construction with piles driven to the bottom pinning the structure to the seabed. These jacket legs and foundation piles were driven at a rake angle of approximately 13° from the vertical. This introduced three-dimensional sound propagation effects as indicated by measurements using a towed array during construction which showed azimuthal variability. In order to model the complicated source, we will use finite element techniques (developed by Sandia National Laboratories) to provide the starting field for a 3D parabolic equation model. Eventually the model predictions will be compared to the actual measurements taken during wind turbine construction. This finite element model will be initially validated by modeling the wave propagation in an instrumented sandbox at the University of Rhode Island (URI) before applying the model to the Block Island modeling scenario. [Work supported by Bureau of Ocean Energy Management (BOEM).]

Exhibit

Exhibit Opening Reception

The instrument and equipment exhibit is located near the registration area in Exhibit Hall D.

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 Sunday, 25 June, 5:30 p.m. to 7:00 p.m., Monday, 26 June, 9:00 a.m. to 5:00 p.m., and Tuesday, 27 June, 9:00 a.m. to 12:00 noon.

Coffee breaks on Monday and Tuesday mornings, as well as an afternoon break on Monday, will be held in the exhibit area.

SUNDAY EVENING, 25 JUNE 2017 ROOM 305, 5:00 P.M. TO 6:00 P.M.

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

C. F. Gaumond, Chair ASC S2
14809 Reserve Road, Accokeek, MD 20607

J. T. Nelson, Vice Chair ASC S2
Wilson Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 108, Mechanical vibration, shock and condition monitoring, and four of its subcommittees, take note—that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Monday, 26 June 2017.

Scope of S2: Standards, specification, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance, and comfort.