

Session 2aIDa**Interdisciplinary: Keynote Lecture****Keynote Introduction—8:00***Invited Paper***8:05**

2aIDa1. Making, mapping, and using acoustic nanobubbles for therapy. Constantin Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, constantin.coussios@eng.ox.ac.uk)

Acoustically driven bubbles continue to find new therapeutic uses, including drug delivery to tumors, opening the blood-brain barrier, and direct fractionation of tissues for surgical applications. Creating acoustic cavitation at length scales and pressure amplitudes compatible with biology remains a major challenge and could be greatly facilitated by a new generation of nano-scale cavitation nuclei that stretch our current understanding of bubble stability, acoustic microstreaming, and the interaction between cavitating bubbles and biological media. Furthermore, in order to ensure patient safety and treatment efficacy, monitoring the location and activity of the nanobubbles during ultrasonic excitation is essential. It will be shown that Passive Acoustic Mapping (PAM) of sources of nonlinear acoustic emissions enables real-time imaging and control of cavitation activity at depth within the body, thereby making it possible to monitor therapy using solely acoustical means. Combined, these techniques for making, mapping, and using acoustic nanobubbles can enable improved delivery of modern immuno-oncology agents to cells and tumors, needle-free transdermal drug delivery and vaccination, and new spinal therapies to treat the spinal cord or repair the intervertebral disc.

Session 2aAAa**Architectural Acoustics: Sound Propagation Modeling and Spatial Audio for Virtual Reality II**

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

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U. Peter Svensson, Cochair

Electronic Systems, Norwegian University of Science and Technology, Acoustics Research Centre, Department of Electronic Systems, Norwegian Univ. of Science and Technology, Trondheim NO - 7491, Norway

Chair's Introduction—9:15*Invited Papers***9:20**

2aAAa1. Fast multipole accelerated boundary element method for the Helmholtz equation in three dimensions on heterogeneous architectures. Nail A. Gumerov (UMIACS, Univ. of Maryland, 115 A.V. Williams Bldg., College Park, MD 20742, gumerov@umiacs.umd.edu) and Ramani Duraiswami (Comput. Sci. & UMIACS, Univ. of Maryland, College Park, MD)

Numerical simulations related to human hearing, architectural and underwater acoustics, and multiple scattering require computational solution of the Helmholtz equation in three dimensions with complex shaped boundaries. Boundary element methods (BEM) are among the most accurate and efficient methods used for this purpose. However, solution of high frequency/large domain problems is

challenging due to poor scaling of conventional solvers with the frequency and domain size. The use of the fast multipole methods (FMM) resolves many problems related to the scalability [N.A. Gumerov and R. Duraiswami, *J. Acoust. Soc. Am.* 125(1), 191205, 2009]. Additional accelerations are needed to be able to solve practical problems over the entire range of human audible frequencies and can be provided using graphics processors (GPUs) with multicore CPUs. In this work, we report development and demonstration of the FMM/GPU accelerated BEM for the Helmholtz equation in 3D designed for hybrid CPU/GPU architectures. Innovations related to choices of preconditioners, parallelization strategies, and choice of optimal parameters will be presented. A single PC version of the algorithm shows accelerations of the order of 10 times compared to the BEM accelerated with the FMM alone. Results for computing head related transfer functions and other standard calculations will be provided.

9:40

2aAAa2. Geometry-based diffraction auralization for real-time applications in environmental noise. Jonas Stienen and Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikustr. 5, Aachen, NRW 52074, Germany, jst@akustik.rwth-aachen.de)

Geometrical acoustics has become the number one choice in a variety of different virtual acoustics applications, especially if physics-based sound field synthesis at real-time rates is desired. The procedures for interactive auralization on the one hand and calculation of room impulse responses as well as outdoor noise prediction on the other hand are converging if not already merely differ in resolution of parameters, i.e., number of rays or order of image sources. When it comes to diffraction effects at geometrical boundaries, however, the discrepancy between model accuracy and calculation efficiency is still high due to the great computational effort required for geometrical path analysis and propagation simulation. The overall goal of ongoing research in this field aims at closing the gap between the two worlds: making diffraction effects audible in a Virtual Reality application with respect to correct physical sound propagation and true reproduction that is required for serious applications in acoustic consulting business and for research, primarily for outdoor noise situations.

10:00

2aAAa3. Plane-wave decomposition and ambisonics output from spherical and nonspherical microphone arrays. Nail A. Gumerov, Dmitry N. Zotkin, and Ramani Duraiswami (VisiSonics Corp., A.V. Williams Bldg. #115, College Park, MD 20742, ramani@umiacs.umd.edu)

Any acoustic sensor embedded in a baffle disturbs the spatial acoustic field to a certain extent, and the recorded field is different from a field that would have existed if the baffle were absent. Recovery of the original (incident) field is a fundamental task in spatial audio. For some sensor baffle geometries, such as the sphere, the disturbance of the field by the sensor can be characterized analytically and its influence can be undone to recover the incident field. However, for arbitrary shaped baffles, numerical methods have to be employed. In the current work, the baffle influence on the field is characterized using boundary element methods, and a framework to recover the incident field from measurements from sensors embedded on the baffle in the plane-wave function basis is developed. Field recovery also allows generation of high-order ambisonics representations of the spatial audio scene. Experimental results using both complex and spherical scatterers will be presented.

10:20–10:40 Break

10:40

2aAAa4. Head-related transfer function personalization for the needs of spatial audio in mixed and virtual reality. Ivan J. Tashev and Hannes Gamper (Microsoft Res. Labs, Microsoft Corp., One Microsoft Way, Redmond, WA 98052, ivantash@microsoft.com)

Virtual Reality (VR) and Mixed reality (MR) devices are typically a head mounted screens, aiming to project objects in an entire environment (VR), or in addition to the existing environment (MR). In both cases, these devices should have the means to create the spatial audio images of the projected objects. Considering the compact form factor of the devices, the most common approach is using binaural audio reproduction through a pair of headphones, isolated (VR) or acoustically transparent (MR). One of the key factors for successful realization of such spatial audio system is personalization of the user Head-Related Transfer Functions (HRTFs). In this paper, we will present and compare several approaches for doing so under the heavy constraints of the VR/MR devices design.

11:00

2aAAa5. Perceptual evaluation on the influence of individualized near-field head-related transfer functions on auditory distance localization. Guangzheng Yu, Yuye Wu, and Bo-sun Xie (School of Phys. and Optoelectronics, South China Univ. of Technol., Wushan Rd. 381#, Tianhe District, Guangzhou, Guangdong 510640, China, scgzyu@scut.edu.cn)

It is desired that spatial audio for virtual reality is able to reproduce various auditory localization information so as to recreate virtual source at different directions and distances. Distance-dependent binaural cue (binaural level difference (ILD), loudness, as well as environmental reflections are considered as auditory distance localization cues. In the case of free and near-field within about 1.0 m, the binaural cue which is encoded in near-field head-related transfer function (HRTFs) is an absolute and dominant distance localization cue [D. S. Brungart, *JASA*, 1999]. However, HRTFs depending on individualized and non-individualized HRTFs are usually used in spatial audio synthesis. In the present work, the perceptual influence of individualized HRTFs distance localization is evaluated by a psychoacoustic experiment. The binaural signals with various bandwidths are synthesized by filtering the input stimuli with individualized and non-individualized near-field HRTFs and then reproduced by headphone. Preliminary results of virtual source localization experiment indicate that individualized HRTFs influences little on distance localization at low frequencies but have some influence at mid and high frequency. [This work was supported by the Natural Science Foundation of China, Grant No. 11574090.]

11:20

2aAAa6. Algebraic reflections in acoustic ray tracing. Erik Molin (Eng. Acoust., Div. of Eng. Acoust., Faculty of Eng., Lund Univ., P.O. Box 118, Lund 22100, Sweden, erik.molin@construction.lth.se)

Stochastic ray tracing is currently one of the most popular geometric acoustic algorithms. It is widely used, and it primarily excels for modeling the late room response at high frequencies. A significant bottleneck of the algorithm is the high computational cost of testing rays for intersection with model geometry. Another is the high number of rays required for convergence. Several methods exist to reduce this cost by means of reducing geometric complexity. This paper proposes a method to algebraically compute transmission paths between receivers and sources by use of bidirectional reflectance distribution functions (BRDF), thus decreasing the number of total rays needed. For each ray-geometry intersection, transmission paths are calculated recursively, and several transmission paths can thereby be considered for each intersection test, while allowing for point-like transmitters and receivers.

11:40

2aAAa7. Three-dimensional remote ensemble system using the immersive auditory display “Sound Cask.” Shiro Ise (School of Information Environment, Tokyo Denki Univ., Muzai Gakuendai Inzai2-1200, Chiba 270-1382, Japan, iseshiro@mail.dendai.ac.jp), Yuko Watanabe (School of Information Environment, Tokyo Denki Univ., Inzai-shi, Japan), and Kanako Ueno (Meiji Univ., Tokyo, Japan)

A 3D sound field simulation system using the immersive auditory display system, “sound cask,” has been developed for creating a virtual environment that would reproduce the 3D acoustics of concert halls for musicians. The simulation system is based on the boundary surface control principle. The original sound field was measured using a microphone array consisting of 80 omnidirectional microphones installed at the nodes of the C80 fullerene structure. The virtual sound field was then constructed in a cask-shaped space (approx. 2×2 m), with 96 channel full-range loudspeakers installed in the space. The 3D acoustic waves of music, including the acoustic condition on the stage, were created virtually inside the sound cask. For this, the first step was to design inverse filters of the MIMO system between the 96 loudspeakers and 80 microphones located in the sound cask. Next, the inverse filters and the impulse responses measured in actual concert halls and signals from instruments played by musicians were convolved in real time. Using two sound casks connected each other, three-dimensional remote ensemble system can be realized. Room acoustic indices between the actual and virtual conditions were compared, and subjective experiments involving professional musicians were performed.

12:00–12:20 Panel Discussion

2a MON. AM

Session 2aAAb**Architectural Acoustics: Acoustic Regulations and Classification of New and Retrofitted Buildings I**

Birgit Rasmussen, Cochair

SBi, Danish Building Research Institute, Aalborg University Copenhagen, A.C. Meyers Vænge 15, Copenhagen SV 2450, Denmark

Jorge Patricio, Cochair

LNEC, Av. do Brasil, 101, Lisbon 1700-066, Portugal

David S. Woolworth, Cochair

*Oxford Acoustics, 356 CR 102, Oxford, MS 38655****Invited Papers*****9:20****2aAAb1. Update on the change in sound insulation requirements in Canada.** Christoph Hoeller and Jeffrey Mahn (Construction, National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, christoph.hoeller@nrc.ca)

In the 2015 edition of the National Building Code of Canada (2015 NBCC, published in January 2016), sound insulation requirements between dwelling units are given in terms of Apparent Sound Transmission Class (ASTC). This is a significant change from the requirements in previous editions of the NBCC which were given in terms of Sound Transmission Class (STC). While the STC rating only accounts for sound transmission through the separating assembly, the ASTC rating also takes into account structural flanking transmission via adjoining building elements. An overview of the change in requirements and of NRC activities to support the code change was given at the ASA meeting in Jacksonville (November 2015). This presentation will provide an update on the implementation of the code change since it came into effect in federal regulations in early 2016. Tools and guidelines provided by the NRC such as Research Report RR-331, "Guide to Calculating Airborne Sound Transmission in Buildings" and the revised version of soundPATHS, NRC's web application to calculate ASTC ratings, will be presented.

9:40**2aAAb2. Accommodation for assemblies in widespread use: The STC 50-ish wall.** Benjamin Markham (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

The classroom acoustics standard ANSI/ASA S12.60 makes a compelling accommodation: a 20 cm (8 in.) concrete masonry unit wall, properly detailed, "...is an acceptable alternate assembly that conforms to the intent of [the requirements]." The relevant requirement is for a Sound Transmission Class (STC) of 50. This wall type is in widespread and generally successful use in U.S. school buildings with concrete construction. An analogous wall in widespread use in steel buildings is a single row of 92 mm (3-5/8 in.) metal studs with two layers of 16 mm (5/8 in.) gypsum board on each side and insulation in the stud cavity. Under certain stud configurations, this wall exceeds STC 50; in the most common stud configurations, however, it falls short. ANSI/ASA S12.60 makes no accommodation for this wall. With requirements for STC 50 walls in FGI guidelines for healthcare, the ANSI/ASA standard for schools, code requirements for multifamily buildings, and in other standards and design guidelines, there is an increasing need to reconcile compliance requirements with constructions commonplace in the United States. This presentation will examine first the acoustical data and then the design implications—both benefits and pitfalls—of accommodations in American standards and design guidelines such as the one made in ANSI/ASA S12.60.

10:00**2aAAb3. Laminated gypsum wallboard in mid-rise wood construction.** Zhiqiang Shi (CertainTeed Acoust. Res., Saint-Gobain Northborough R&D Ctr., 9 Goddard Rd., Northborough, MA 01532, zhiqiang.shi@saint-gobain.com) and Robert Marshall (CertainTeed Bldg. Sci., Saint-Gobain, Mississauga, ON, Canada)

Mid-rise wood construction is a cost-effective and sustainable choice to achieve high performance in commercial and multi-family residential housing. It is gaining popularity in the industry as several building codes, including IBC (2009) and NBCC (2015), now allow five- and six-story constructions, respectively. Acoustic comfort in such buildings is important to attract and retain occupants. Because of its lightweight construction nature, the new code changes the requirements for airborne sound insulation between dwellings from a Sound Transmission Class (STC) rating which only describes the sound insulation of the common partition between rooms to an ASTC rating, which includes contributions from all of the flanking paths. This paper shows laminated SilentFX/QC gypsum wall board is a fast, economical, space-saving, and code-compliant solution in meeting the ASTC requirement of the new building codes with examples.

The test methodology is based on ISO 15712 and ISO 10848 standard, and follows the procedure outlined in NRC-331 publication. It consists of testing STC and radiation efficiency of wall partitions and measuring the vibration reduction index of various conjunctions between the partitions and different flanking paths. The ASTC predictions were validated with prior results obtained from the NRC Flanking Facility.

10:20

2aAAb4. Comparison of Sound Transmission Class and Outdoor-Indoor Transmission Class for specification of exterior facade assemblies. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

Design and specifications of the acoustical performance of the exterior facade of buildings to protect the occupants from transportation noise sources is a common acoustical task. Single number quantities (SNQ) are useful to specify and communicate acoustical performance to other design professionals. Common SNQ values include Sound Transmission Class (STC) and Outdoor-Indoor Transmission Class (OITC). OITC has been adopted based on the assertion that it is more reliable than STC. This paper will compare the SNQ using a large dataset and evaluate if there is any advantage of using OITC over STC as the method of specifying or communicating exterior facade performance.

10:40–11:00 Break

11:00

2aAAb5. Brazilian acoustics regulation: An overview and proposal of façade requirements. Carolina R. Monteiro, Marcel Borin (Res. and Development, Harmonia Acústica Davi Akkerman + Holtz, Av. Mofarrej 1200, São Paulo, São Paulo 053111000, Brazil, carolina.monteiro@harmoniaacustica.com.br), Mariana Shimote, Teddy Yanagiya (Project Management, Harmonia Acústica Davi Akkerman + Holtz, São Paulo, São Paulo, Brazil), and María Machimbarrena (Appl. Phys., Universidad de Valladolid, Valladolid, Valladolid, Spain)

The Brazilian acoustics regulation for new residential buildings entered into force in 2013, and during the past years, the construction and acoustic consultancy market developed new procedures to accomplish with the façade requirements. In this paper, typical existing methods are presented, as well as new proposals to be incorporated in the future revision of the Regulation. Furthermore, existing and possible future Brazilian requirements are translated to the suggested descriptor $D_{nT,50}$ and compared within the acoustic classification scheme for dwellings proposed in ISO/CD 19488.

11:20

2aAAb6. Improvement of acoustic and thermal performances of façades on retrofit buildings. Giovanni Semprini (Dept. of Industrial Eng., Univ. of Bologna, Viale Risorgimento 2, Bologna 40136, Italy, giovanni.semprini@unibo.it), Antonino Di Bella (Dept. of Industrial Eng., Univ. of Padova, Padova, PD, Italy), Simone Secchi (Dept. of Industrial Eng., Univ. of Florence, Firenze, Italy), Luca Barbaresi (Dept. of Industrial Eng., Univ. of Bologna, Bologna, Bologna, Italy), Nicola Granzotto (Dept. of Industrial Eng., Univ. of Padova, Padova, Italy), and Anastasia Fotopoulou (Dept. of Architecture, Univ. of Bologna, Athens, Greece)

Deep renovation of existing buildings represents an important strategy to go beyond the standard energy retrofit (as the increasing of thermal insulation of building envelope) and to provide more benefits to residents considering non-energetic aspects, like the improvement of architectural quality of building and of indoor life quality. In this context, façade's additional elements (balconies, solar greenhouses, etc.) can be effective solutions in order to increase thermal and acoustic performances as well as providing a more living space to the outside. This paper analyzes and compares different technical construction systems for façade improvement in order to define robust solutions correlated both to outdoor climate context (to increase energy performances towards nearly Zero Energy Buildings) and to outdoor noise for a better acoustic sound insulation. A case study is presented, and dynamic energy simulations, performed on a building made with materials and construction technology widespread in Italy in the '60s, show the energy impact of different solutions. The evaluation of acoustic façade insulation highlights the need to consider not only the properties of materials but also the effects of flanking transmission, sometimes correlated to thermal bridges, and façade shapes of which continuous balcony solutions give an important increase on acoustic performances.

11:40

2aAAb7. Meta-analysis of subjective-objective sound insulation data. Jonas Brunskog (Acoust. Technol., DTU Elec. Eng., Elektrovej, Bldg. 352, Kgs. Lyngby DK-2800, Denmark, jbr@elektro.dtu.dk) and Birgit Rasmussen (Aalborg University Copenhagen (AAU-CPH), SBI, Danish Bldg. Res. Inst., Copenhagen, Denmark)

A smaller meta-analysis using field data for subjective sound insulation in dwellings is carried out, using and analyzing data found in the literature. The investigated parameter is the correlation coefficient between the sound insulation metrics and the subjective annoyance (or similar). Both airborne and impact sound insulation is considered. One of the objectives are to see if low frequency adaptation terms according to ISO 717 will yield an increased correlation or not. Other investigated aspects is the influence of lightweight versus heavyweight building elements and the question regarding the importance of vertical versus horizontal transmission.

12:00

2aAAb8. On the definition of acoustic comfort in residential buildings.

Nikolaos - Georgios Vardaxis, Delphine Bard (Construction Sci., Lund Univ., John Ericssons väg 1, Lund 22100, Sweden, nikolas.vardaxis@construction.lth.se), and Kerstin Persson Waye (Occupational and Environ. Medicine, Sahlgrenska Acad., Gothenburg Univ., Gothenburg, Sweden)

The aim of this study is to explore acoustic comfort in family apartments in Scandinavia, not only regarding standardized acoustic data but also including the users' perception of their living sound environment. A first

approach takes place for the definition of the concept of acoustic comfort, which was rarely defined before and used to be mostly interpreted as absence, low, or acceptable levels of noise in a place. In this article, acoustic comfort is not only lack of discomfort, it is explained as the ability to have proper sound conditions in overall, for a certain activity in a certain space, considering physical characteristics and the users' demands. Then, a method is set up for the evaluation of comfort in dwellings, including acoustic measurements and social surveys in test buildings. A questionnaire for the collection of the subjective responses of the tenants, regarding noise annoyance, sound perception, and emotions, is presented and analyzed.

MONDAY MORNING, 26 JUNE 2017

ROOM 206, 9:20 A.M. TO 12:20 P.M.

Session 2aAac

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

Arianna Astolfi, Cochair

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Viveka Lyberg-éhlander, 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

9:20

2aAAc1. Teachers' voice parameters and classroom acoustics—A field study and online survey. Nick Durup, Bridget M. Shield, Stephen Dance (London South Bank Univ., LSBU, 103 Borough Rd., London SE1 0AA, United Kingdom, nicksenate@hotmail.com), and Rory Sullivan (Sharps Redmore Acoust. Consultants, Ipswich, United Kingdom)

Many studies have suggested that teachers have a significantly higher rate of voice problems than the general population. In order to better understand the possible influences of room acoustics on different voice parameters, a study has been carried out by London South Bank University which involved measurements of voice parameters for teachers working in classrooms with varying acoustic conditions. Data relating to the voice, including the average speech sound level, fundamental frequency, and phonation percentage, were captured using an Ambulatory Phonation Monitor (APM) which measured directly from skin vibrations in the neck, thereby excluding the effects of other noise sources in the environment. The measured voice parameters were compared with the room acoustic data for the classrooms involved, which were surveyed separately from the voice measurements. In addition to the field measurements, an online questionnaire was undertaken with the support of two UK teacher trade unions. This was designed to gain further information on teachers' experiences of voice problems and school acoustics in general and indicated that over 66% of the surveyed teachers had experienced voice problems during their career. This paper will present the results of the field measurements and questionnaire survey.

2aAAc2. Long-term voice monitoring with smartphone applications and contact microphone. Arianna Astolfi (Dept. of Energy, Politecnico di Torino, Corso DC degli Abruzzi, 24, Turin 10124, Italy, arianna.astolfi@polito.it), Alessio Carullo, Simone Corbellini (Dept. of Electronics and Telecommunications, Politecnico di Torino, Turin, Italy), Massimo Spadola, Anna Accornero (Dept. of Surgical Sci., A.O.U. Città della Salute e della Scienza di Torino, Turin, Italy), Giuseppina E. Puglisi (Dept. of Energy, Politecnico di Torino, Torino, Italy), Antonella Castellana (Dept. of Electronics and Telecommunications, Politecnico di Torino, Turin, Italy), Louena Shtrepi (Dept. of Energy, Politecnico di Torino, Torino, Italy), Gian Luca D'Antonio (Dept. of Management and Production Eng., Politecnico di Torino, Turin, Italy), Alessandro Peretti (School of Specialization in Occupational Medicine, Università di Padova, Padova, Italy), Giorgio Marcuzzo, Alberta Pierobon, and Giovanni B. Bartolucci (Dept. of Cardiac, Thoracic and Vascular Sci., Università di Padova, Turin, Italy)

In recent years, the growing interest in the recognition of voice disorders as occupational diseases has required screening methods adaptable to the clinical requirements, capable to extend the collection of baseline data. In this framework, the use of smartphones has gained increasing interest, thanks to advancements in digital technology, which made them suitable for recording and analyzing acoustic signals. Two smartphone applications, based on the Voice Care® technology, have been developed for long-term monitoring of voice activity when combined with a cheap contact microphone embedded in a collar. The applications have been tested in laboratory and used for the monitoring of teachers at kindergarten, primary school, and university. *Vocal Holter App* allows the selection of short and long term monitoring mode, and three different clusters of vocal parameters related to intensity, intonation, and load, respectively. Most of the results are based on the distributions of occurrences of vocal parameters. A highlight informs the person under monitoring of pathologic voice. *Vocal Holter Rec* allows data recording and to perform a personalized analysis based on updated parameters. The equipment allows downloading and saving data on a dedicated web site for further processing, comparisons over time, or sharing with physicians or rehabilitators.

10:00

2aAAc3. Vocal fatigue in virtual acoustics scenarios. Pasquale Bottalico (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Lansing, MI 48910, pb@msu.edu), Lady Catherine Cantor Cutiva, and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

The overuse of the voice by professional voice users, such as teachers, is known to cause physiological vocal fatigue. Vocal fatigue is used to denote negative vocal adaptation that occurs as a consequence of prolonged voice use or vocal load. This study investigates how self-reported vocal fatigue is related to voice parameters (sound pressure level SPL, fundamental frequency f_0 , and their standard deviations) and the duration of the vocal load. Thirty-nine subjects were recorded while reading a text. Different acoustics scenarios were artificially created to increase the variability in the speech produced (3 reverberation time, 2 noise conditions, and 3 auditory feedback levels), for a total of 18 tasks per subject presented in a random order. For each scenario, the subjects answered questions addressing their perception of vocal fatigue on a visual analog scale. A model of the vocal fatigue to acoustic vocal parameters is proposed. The duration of the vocal load contributed to 55% of the variance explained by the model, followed by the interaction between the standard deviations of the SPL and f_0 (24%). The results can be used to give a simple feedback during voice dosimetry measurements.

10:20

2aAAc4. Individual factors and its association with experienced noise annoyance in Swedish preschools. Fredrik Sjödin (Psych., Umeå Univ., Beteendevetarhuset, Umeå 90187, Sweden, fredrik.sjodin@umu.se)

Studies have shown that preschool teachers often report having a troublesome working environment in terms of high noise levels. Noise annoyance is often reported from employees working under poor acoustical conditions. The aim of the study was to investigate whether there is an association between rated noise annoyance and actual noise exposure for Swedish preschool teachers. Furthermore, the study also aimed to investigate whether preschool teacher with different individual characteristics differs in their rated noise annoyance at work. The study included 90 preschool teachers in Sweden. Data were collected by use of personal carried noise dosimeters and by questionnaires during one representative work week. The average equivalent noise exposure was 71 dBA and the average rated noise annoyance was 65 on a 0-100 mm scale. Rated noise annoyance was not correlated to the sound exposure during a work week ($r = 0.66$, $P = 0.42$). Analysis of differences in noise annoyance ratings between preschool teachers with different individual characteristics (hearing impairment, tinnitus, age, and gender) revealed no significant statistical group differences. Other factors needs to be investigated to better explain what affects differences in rated noise annoyance at work among preschool teachers in Sweden.

10:40

2aAAc5. One-year longitudinal study on teachers' voice parameters in secondary-school classrooms: Relationships with voice quality assessed by perceptual analysis and voice objective measures. Antonella Castellana (Dept. of Electronics and Telecommunications, Politecnico di Torino, Corso DC degli Abruzzi, 24, Turin, TO 10129, Italy, antonella.castellana@polito.it), Giuseppina E. Puglisi, Giulia Calosso (Dept. of Energy, Politecnico di Torino, Torino, TO, Italy), Anna Accornero (Dept. of Surgical Sci., Università degli Studi di Torino, Turin, Italy), Lady Catherine Cantor Cutiva (Dept. of Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), Fiammetta Fanari (Dept. of Surgical Sci., Università degli Studi di Torino, Torino, TO, Italy), Franco Pellerey (Dept. of Mathematical Sci., Politecnico di Torino, Torino, TO, Italy), Alessio Carullo (Dept. of Electronics and Telecommunications, Politecnico di Torino, Turin, Italy), and Arianna Astolfi (Dept. of Energy, Politecnico di Torino, Turin, Italy)

This longitudinal work explores the relationships between three analyses used for assessing teachers' voice use: the voice monitoring during lessons that describes the teachers' Vocal Behavior (VB), the perceptual assessment of voice by speech-language pathologists and the estimation of objective parameters from vocalizations to define teachers' Vocal Performance (VP). About 30 Italian teachers from secondary schools were involved at the beginning and at the end of a school year. In each period, teachers' vocal activity was monitored using the Voice Care device, which acquires the voice signal through a contact microphone fixed at the neck to estimate sound pressure level, fundamental frequency, and voicing time percentage. Once in each period, two speech-language pathologists performed a perceptual assessment of teachers' voice using the GIRBAS-scale. On that occasion, teachers vocalized a sustained vowel

standing in front of a sound level meter in a quiet room. Jitter, Shimmer and other parameters were extracted using Praat, while a new metric of Cepstral-Peak-Prominence-Smoothed was estimated with a MATLAB script. Several relationships between the outcomes of each analysis were investigated, e.g., statistical differences between the dimension "G" from GIRBAS-scale and objective measures for VB and VP, and correlations between objective measures and perceptual ratings were assessed.

11:00

2aAAc6. Speech sound pressure level distributions and their descriptive statistics in successive readings for reliable voice monitoring. Antonella Castellana, Alessio Carullo (Dept. of Electronics and Telecommunications, Politecnico di Torino, Corso DC degli Abruzzi, 24, Turin, TO 10129, Italy, antonella.castellana@polito.it), Umberto Fugiglando (Senseable City Lab, Massachusetts Inst. of Technol., Cambridge, MA), Giuseppina E. Puglisi (Dept. of Energy, Politecnico di Torino, Torino, Italy), and Arianna Astolfi (Dept. of Energy, Politecnico di Torino, Turin, Italy)

Due to the high prevalence of voice disorders among teachers, there is a growing interest in monitoring voice during lessons. However, the reliability of the results is still to be deepened, especially in the case of repeated monitoring. The present study thus investigates the speech Sound Pressure Levels (SPL) variability under repeatability conditions aiming to provide preliminary normative data for the results assessment. In a semi-anechoic chamber, 17 subjects read twice and subsequently two phonetically balanced passages, which were simultaneously recorded with a sound level meter, a headworn microphone, and a portable vocal analyzer. Each speech sample was characterized through the distribution of SPL occurrences and several descriptive statistics of SPL distribution were calculated. For each subject, statistical differences between the two SPL distributions related to each passage were investigated using the Mann-Whitney U-test. For each group of subjects using the same device, the Wilcoxon signed-rank test was applied to the paired lists of descriptive statistics related to each passage. For mean, mode, and equivalent SPL, the within-speaker and the within-group variability were assessed for each device. For all the devices and SPL parameters, the within-speaker variability was not higher than 2 dB while the within-group variability reached 5.3 dB.

Contributed Papers

11:20

2aAAc7. Vocal effort of teachers in different classrooms. Jamilla Balint, Rafael P. Ludwig, and Gerhard Graber (Signal Processing and Speech Commun. Lab., Graz Univ. of Technol., Inffeldgasse 16c, Graz, Styria 8010, Austria, balint@tugraz.at)

Classroom acoustics can be one of the reasons for causing voice disorders among teachers. The purpose of this study was to investigate the vocal effort of teachers and the noise level of students under different classroom conditions (reverberant and less reverberant classrooms, junior and senior level, different teaching types like teacher-centered teaching and team work). We were able to carry out measurements over a period of two weeks, where the same two classes were taught by the same teachers in a reverberant classroom during the first week and in a less reverberant classroom during the second week. The results indicate a much greater increase in the vocal effort of the teachers over a period of 6 hours in a reverberant classroom compared to a less reverberant classroom. In addition, teaching younger students requires a greater vocal effort since the noise level is much higher. Also, extended breaks are crucial for the recovery of the voice in reverberant spaces, whereas the voice needs less time to recover in less reverberant spaces. A questionnaire filled out by the participating teachers confirmed that the level of exhaustion at the end of the day is much greater in reverberant spaces.

11:40

2aAAc8. Norwegian experiences of acoustics in classrooms. Anders Homb (SINTEF Bldg. & Infrastructure, Høgskoleringen 7B, Trondheim 7465, Norway, anders.homb@sintef.no)

During the last decades, the investments on new school buildings and retrofitting of existing buildings have been considerable in Norway and it will continue for the next years. A characteristic of this period have also been a development of the teaching methods and as a consequence of that, a change of the area planning and room layout, in the beginning without considering the acoustic challenges sufficiently. The acoustical norms,

guidelines, and recommendations for schools were revised afterwards. Important issues in these documents were on reverberation time, speech intelligibility, and on background noise necessary for sufficient conditions during the educational process. The paper will present the development of the acoustical requirements and recommendations in this period together with some layout examples. The revised requirements and recommendations have been based on international experiences and some Norwegian research studies. The paper will present some results from these studies, both from measurements of relevant parameters and evaluation of student perception. The analysis will focus on the reverberation time, signal to noise ratio, and how to prevent noise in the classroom area from adjacent spaces. Finally, the paper will give some suggestions on how to improve the acoustical conditions in the spaces for the future.

12:00

2aAAc9. Acoustical measurement of modern Japanese libraries. Kazuma Shamoto, Mai Ikawa, Hiroshi Itsumura (Univ. of Tsukuba, 1-2 Kasuga, Tsukuba, Ibaraki 305-8550, Japan, shamoto.kazuma.ru@alumni.tsukuba.ac.jp), Koji Ishida (Ono Sokki, Yokohama, Japan), and Hiroko Terasawa (Univ. of Tsukuba, Tsukuba, Ibaraki, Japan)

The function of modern libraries is transforming from a quiet reading space to an active, social, and interactive learning place for old and young people including children. We investigate the room acoustics of modern libraries in order to examine if they can accommodate both needs of silence and bustling communication. We measured impulse responses and sound-decay distribution patterns at three libraries (two university libraries and a public library) with different architectural styles. In addition, we measured the noise level during the library opening hours with active visitors. Every library showed different patterns of sound propagation and visitor activities. Library spaces with densely installed shelves highly absorb noises, while spaces with few shelves and a multiple-height structure were highly echoic. Some carefully designed spaces showed a clear acoustical zoning, that the sound from bustling noise area hardly reach to the quiet reading area even with a high ceiling and sparse shelves.

Session 2aAB

Animal Bioacoustics: Behavior/Comparative Studies

Peter L. Tyack, Chair

Biology, University of St. Andrews, Sea Mammal Research Unit, Scottish Oceans Institute, St. Andrews KY16 8LB,
United Kingdom

Contributed Papers

9:20

2aAB1. Shipboard echosounders negatively affect acoustic detection rates of beaked whales. Danielle Cholewiak, Annamaria Izzi DeAngelis, Peter Corkeron, and Sofie M. Van Parijs (NOAA Northeast Fisheries Sci. Ctr., 166 Water St., Woods Hole, MA 02543, danielle.cholewiak@noaa.gov)

Beaked whales are cryptic, deep-diving odontocetes that are sensitive to anthropogenic noise. While their behavioral responses to navy sonar have been the subject of extensive study, little effort has been expended to evaluate their responses to other types of acoustic signals, such as fisheries echosounders. From 1 July to 10 August 2013, the Northeast Fisheries Science Center conducted a shipboard cetacean assessment survey, combining visual observation and passive acoustic data collection. Simrad EK60 echosounders were used to collect prey field data; echosounder use was alternated on/off on a daily basis to test for an effect on beaked whale detection rates. The software package Panguard was used to detect, classify, and localize individual beaked whales. A GLM was used to test the relationship between acoustic detections and covariates; echosounder use negatively affected beaked whale acoustic detection rates, and acoustic event durations were significantly shorter. These results suggest that beaked whales are reacting to echosounders by either moving away or interrupting foraging activity. This decrease in detectability has implications for management and mitigation activities. The use of scientific echosounders is rapidly increasing, thus leading to potentially broad ecological implications for disturbance effects on these sensitive species as well.

9:40

2aAB2. Signature whistles facilitate reunions and/or advertise identity in Bottlenose Dolphins. Nicholas Macfarlane (Int. Union for the Conservation of Nature, Washington, DC), Vincent Janik (Biology, Univ. of St. Andrews, St. Andrews, Fife, United Kingdom), Frants H. Jensen (Aarhus Inst. of Adv. Studies, Aarhus Univ., Aarhus, Denmark), Katherine McHugh (Chicago Zoological Society's Sarasota Dolphin Res. Program, Mote Marine Lab., Sarasota, FL), Laela Sayigh (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA), Randall Wells (Chicago Zoological Society's Sarasota Dolphin Res. Program, Mote Marine Lab., Sarasota, FL), and Peter L. Tyack (Biology, Univ. of St. Andrews, Sea Mammal Res. Unit, Scottish Oceans Inst., East Sands, St. Andrews, Fife KY16 8LB, United Kingdom, plt@st-andrews.ac.uk)

Animals with stable relationships need mechanisms to stay in touch when separated. Five decades of research suggest that signature whistles are likely candidates for serving this contact-calling purpose in bottlenose dolphins. However, difficulties identifying the vocalizing individual and measuring inter-animal distances have hindered tests of call functions among wild dolphins. Moreover, signature whistles almost certainly serve a variety of functions, so that to focus on contact calling, it is useful to identify contexts where animals need to maintain cohesion. By simultaneously tagging pairs of mothers and calves to look at instances when the animals are separating and reuniting, we focus on testing specific contact functions of signature whistles. Drawing from the literature, we define three potential contact

call functions for signature whistles, each with its own hypothetical signature whistle distribution during separations and reunions: location monitoring, reunion calls, and identity advertisement calls. To test these potential functions, we estimated the probability of an animal producing a signature whistle at different stages of temporary separation events. Using a binomial logistic regression model, we found that the data are consistent with signature whistles functioning as reunion calls or identity advertisement calls but not for location monitoring during separations and reunions.

10:00

2aAB3. First harmonic shape analysis of Brazilian free-tailed bat calls during emergence. Yanqing Fu and Laura Kloepper (Dept. of Biology, Saint Mary's College, 264 Sci. Hall, Notre Dame, IN 46556, yfu@saintmarys.edu)

Echolocating bats can adapt calls when facing challenging echolocation tasks. Previous studies have shown that bats can change their pulse duration, pulse repetition rate, or vary their start/end/peak frequencies depending on behavior. Even though this kind of signal investigation reveals important findings, these approaches to analysis use bulk parameters that may hide subtleties in the call structure that could be important to the bat. In some cases, calls may have the same start and end frequencies but have different FM shapes and meet different sensory needs. In the present study, we demonstrate an algorithm for extracting the first harmonics of the Brazilian free-tailed bat (*Tadarida brasiliensis*) to investigate how the shape of the call changes. High pass filtering, power banded time-frequency analysis, and search algorithms were used to isolate the first harmonics. By tracking the first harmonics, the detailed frequency modulation shapes of different bat group sizes were obtained, and the difference among those traces was measured. The detailed shape analysis will provide a new insight into the adaptive call design of bats.

10:20–10:40 Break

10:40

2aAB4. Vocalization in clouded leopards and tigers: Further evidence for a proximate common ancestor. Edward J. Walsh (Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, edward.walsh@boystown.org), Heather Robertson, Sandy Skeba (Nashville Zoo at Grassmere, Nashville, TN), and JoAnn McGee (Boys Town National Res. Hospital, Omaha, NE)

Previously, we reported that clouded leopards (*Neofelis nebulosa*) share an unusual auditory response timing characteristic with tigers (*Panthera tigris*) and possibly other representatives of the genus *Panthera*. Peripheral auditory response timing, or latency, and stimulus frequency relationships obey a rule in these species that is more parabolic in nature than the commonly observed inverse rule. That is, neural response latencies to low frequencies are as short, or shorter, than those measured at higher frequencies in the measurable response spectrum. In this report, we consider the possibility that clouded leopards may share other bioacoustic qualities that differentiate large roaring cats from smaller members of the cat family, Felidae.

In that context, vocalizations were recorded from young adult clouded leopards at the Nashville Zoo at Grassmere and distinctive similarities within a subset of *P. tigris* calls were observed. The acoustic properties of the low level, friendly greeting known as prusten, more commonly known as chuffing, were of particular interest. Although similar in nature, quantitative and structural differences clearly differentiate the intraspecies call and differences may provide insight to production mechanisms. This observation is of particular interest in an evolutionary context because *N. nebulosa* and *P. tigris* shared a proximate common ancestor.

11:00

2aAB5. A first description of rhythmic song in Omura's whale (*Balaenoptera omurai*). Salvatore Cerchio (New England Aquarium, Central Wharf, Boston, MA 02110, scerchio@gmail.com), Sandra Dorning (Univ. of Oregon, Eugene, OR), Boris Andrianantenaina (Les Baleines Asseau, Nosy Be, Madagascar), and Danielle Cholewiak (NOAA Northeast Fisheries, Woods Hole, MA)

Omura's whale is a recently described tropical Balaenopterid whale with virtually nothing known about their acoustic behavior. Recordings have revealed a stereotyped 15-50 Hz amplitude-modulated vocalization, rhythmically repeated in a typical *Balaenoptera* song manner. In order to describe the characteristics of the song, continuous recordings were made using archival recorders during 21 days at 4 sites off the northwest coast of Madagascar in documented Omura's whale habitat. A total of 926 hours of recordings were manually browsed to identify all occurrences of the song vocalizations, logging 9117 individual song units. Occurrence varied among sites spread across 40 km of shelf habitat, indicating heterogeneous distribution of whales and use of habitat over space and days. Diel variation indicated higher incidence of song during daylight hours, counter to trends found in other Balaenopterid whales. A total of 215 different individual series were identified ranging from 3 to 252 consecutive song units. For 121 individuals with more than 20 consecutive song units, the interval ranged from 147.4 s to 289.0 s with a mean of 200.3 s (s.d. 25.9), and recorded song duration ranged up to 13.33 hr. This represents the first description of singing behavior for this species, suggesting a time-intensive behavioral display likely related to breeding.

11:20

2aAB6. Two new whale calls in the southern Indian Ocean, and their geographic and seasonal distribution over five sites and seven years. Emmanuelle C. Leroy (Laboratoire GeoSci. Ocean, Univ. of Brest, IUEM Technopole Brest Iroise, Rue Dumont d'Urville, Plouzané 29280, France, emmanuelle.leroy@univ-brest.fr), Flore Samaran, Julien Bonnel (Lab-STICC, ENSTA Bretagne, Brest cedex 9, France), and Jean-Yves Royer (Laboratoire GeoSci. Ocean, Univ. of Brest, Plouzané, France)

Since passive acoustic is widely used to monitor cetaceans, unidentified signals from biological sources are commonly reported. The signal's characteristics and emission patterns could give keys to identify the possible sources. Here, we report two previously unidentified signals found in acoustic records from five widely spread sites in the southern Indian Ocean and spanning seven years (2007, 2010 to 2015). The first reported signal (M-call) consists of a single tonal unit near 22 Hz and lasting about 10 s. The second signal (P-call) is also a tonal unit lasting about 10 s, but at a frequency near 27 Hz. The latter has often been interpreted as an incomplete Antarctic blue whale Z-call (*Balaenoptera musculus intermedia*). From a systematic analysis of our acoustic database, we show that both signals have similar characteristics as blue whale vocalizations, but with spatial and seasonal patterns that do not resemble any of the known populations dwelling in the southern Indian Ocean. M-calls are recorded only in 2007, while P-calls are present every recording year, with an increasing abundance with time. P-calls may co-occur with but are clearly distinct from Z-calls. The sources of the two new calls have yet to be visually identified.

11:40

2aAB7. The effects of wind turbine wake turbulence on bat lungs. Dorien O. Villafranco, Sheryl Grace, and Ray Holt (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, dvillafr@bu.edu)

Bat mortality is known to increase near wind turbines. Recent studies are in disagreement as to the exact cause of death of these bats. Literature suggests that they are either killed upon direct contact with the turbine blades or by barotrauma. In barotrauma, a sudden change in the surrounding air-pressure causes tissue damage in biological structures that contain air, most notably the lungs. The present work develops a computational model of the bat lung, in which the lung is modeled as a gas bubble with an elastic shell immersed in a fluid, whose dynamics are governed by a Rayleigh-Plesset-like equation. Pressure gradients near the wind turbine are obtained using computational fluid dynamics. The lung's response to pressure changes is attained by simulating the pressure's effect on the gas bubble. The study allows for a greater understanding of bat barotrauma and its potential link to wind turbine pressure fields.

Session 2aAO**Acoustical Oceanography: Session in Honor of David Farmer I**

Tim Leighton, Cochair

Institute of Sound and Vibration Research, University of Southampton, Southampton, United Kingdom

Grant B. Deane, Cochair

Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St., La Jolla, CA 92093-0238

Andone C. Lavery, Cochair

*Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 98 Water Street, MS 11, Bigelow 211, Woods Hole, MA 02536***Chair's Introduction—9:15*****Invited Papers*****9:20****2aAO1. What's in the water? Remote sensing of currents and "stuff" in the ocean.** Gregory B. Crawford (Faculty of Sci., Univ. of ON Inst. of Technol., 2000 Simcoe St. N., Oshawa, ON L1H 7K4, Canada, greg.crawford@uoit.ca)

High frequency acoustic remote sensing of the ocean was still in its infancy in the early 1980s. This paper reviews some early collaborative work with, and subsequent studies informed and inspired by Dr. David Farmer. We summarize early efforts to assess ocean currents, turbulence, and air-sea gas exchange. We will also review a few somewhat serendipitous research studies involving the acoustic assessment of fish wakes and sand dollar populations. More recent results associated with the measurement of tsunami currents will be presented, which provide a gentle reminder to those who would use acoustic velocity estimates to ask: what's in the water?

9:40**2aAO2. Submesoscale dynamics in the coastal ocean.** Burkard Baschek, Ingrid Benavides, Ryan P. North (Inst. of Coastal Res., Helmholtz-Zentrum Geesthacht, Max-Planck-Str. 1, Geesthacht 21502, Germany, Burkard.Baschek@hzg.de), Geoffrey Smith, and Dave Miller (Naval Res. Lab., Washington, DC)

High-resolution observations reveal the fast dynamics of submesoscale eddies in the coastal ocean. The eddies seem to play an important role in the ocean energy cascade and are thought to be important drivers for phytoplankton production. The eddies are frequently observed in the coastal and open ocean and are characterized by sharp gradients of $1^{\circ}\text{C}/\text{m}$ and high Rossby numbers >10 . In order to simultaneously resolve the short temporal and spatial scales of submesoscale eddies, an observational multi-platform approach with planes, a zeppelin, several vessels, gliders, and floats was used yielding a horizontal and vertical resolution of $<1\text{ m}$ with repeat observations every 1 to 15 min. The Submesoscale Experiments (SubEx) took place off Catalina Island, CA, and off Bornholm in the Baltic Sea. Observations were carried out with aerial sea surface temperature and hyperspectral measurements, rapid in situ measurements with a towed instrument array, gliders with turbulence probes, as well as surface and subsurface velocity measurements with drifters, as well as Radar and Acoustic Doppler Current Profilers. Additional SAR, SST, and ocean color satellite imagery is used to investigate the occurrence of submesoscale eddies in the coastal ocean. The observations indicate intense mixing, turbulent dissipation, and subsequent stratification. The temperature distribution is closely linked to phytoplankton concentrations suggesting a strong bio-physical coupling.

10:00**2aAO3. Imaging and mapping water mass intrusions and internal waves.** Timothy F. Duda, Andone C. Lavery, and Glen Gawarkiewicz (Woods Hole Oceanographic Inst., WHOI APOE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu)

Oceanic intrusions and internal waves can each be imaged with echo sounders for dynamics and flux studies. We have recently investigated signatures of intrusions, while internal-wave imaging is long established, with Farmer as a leader. Intrusive flow at fronts with counteracting temperature and salinity gradients along isopycnals is one of many mixing phenomena in the ocean. Advection within the intrusions is accompanied by diapycnal mixing above and below that may provide buoyant forcing. Probable double-diffusive mixing processes can create sharp interfaces and microstructure that can provide intrusion-following echo patterns. This backscattering is weaker than that from shear-generated turbulence microstructure, but it can be spatially coherent along intrusions and thus robustly detectable. Mapping of intrusion features with a shipboard system using this method may enable targeted physical measurements within intrusions to advance our knowledge, and can provide 2D structure maps if augmenting dropped probes are used. Note that plankton

backscattering may mask these signals in some environments and seasons. Internal-wave imaging by recording the geometry of passive tracers moved vertically by the wave motions can be accomplished with even modest narrowband systems in plankton-rich environments, as shown with Quantifying, Predicting, and Exploiting Uncertainty program data from the East China Sea.

10:20–10:40 Break

10:40

2aAO4. Recent development and application of inverted echo sounders in observational physical oceanography. Qiang Li (Graduate School at Shenzhen, Tsinghua Univ., B101 Tsinghua Campus, Nanshan Xili University Town, Shenzhen 518055, China, li.qiang@sz.tsinghua.edu.cn), David Farmer (Inha Univ., Vancouver, Br. Columbia, Canada), Timothy F. Duda (Woods Hole Oceanographic Inst., Woods Hole, MA), Steven R. Ramp (Monterey Bay Aquarium Res. Inst., Carmel Valley, CA), Xianzhong Mao (Graduate School at Shenzhen, Tsinghua Univ., Shenzhen, China), Jae-Hun Park (Inha Univ., Seoul, South Korea), and Xiao-Hua Zhu (The Second Inst. of Oceanogr., Hangzhou, China)

Since the first experiment carried out by Rossby in Bermuda, inverted echo sounders have been used in physical oceanography observations for half a century. The inverted echo sounder measures the round-trip acoustic travel time from the sea floor to the sea surface, thus acquiring vertically integrated information on the thermal structure, from which the first baroclinic mode of thermocline motion can be inferred. Arrays of inverted echo sounders have been deployed almost all over the global ocean to observe internal waves, mesoscale eddies, western boundary currents, etc., providing valuable targeted data for physical oceanographers. Acoustic aspects of the inverted echo sounder performance have been recently examined. Sources of error affecting instrument performance include tidal effects, barotropic adjustment, ambient acoustic noise and sea surface roughness. The latter two effects are explored with a simulation that includes surface wave reconstruction, acoustic scattering based on the Kirchhoff approximation, wind generated noise, sound propagation, and the instrument's signal processing circuitry. Not only does the analysis enhance our understanding of the acoustic travel time data but also suggests new approaches to extend the application of inverted echo sounders, for example, for sensing wind over the sea. New deployments of inverted echo sounders and recent developments in hardware system and auxiliary accessories are also introduced.

11:00

2aAO5. Observations of Langmuir circulation in the open ocean using acoustic instrumentation mounted on a subsurface drifting package. Len Zedel (Phys. and Physical Oceanogr., Memorial Univ. of NF, Chemistry-Phys. Bldg., Memorial University, St. John's, NF A1B 3X7, Canada, zedel@mun.ca)

The air-sea interface of the ocean is an energetic and dynamic environment that poses challenges for the positioning of instrumentation. One way to avoid the complexities of placing instrumentation at the ocean surface is to position a platform at some distance below the surface and sample surface processes remotely using acoustic systems. David Farmer recognized the value of such a sampling approach and led the Ocean Acoustics group at the Institute of Ocean Sciences to develop the required capabilities in the late 1980s. We report on observations made during a cruise to Ocean Station Papa in October 1987. Acoustic instruments revealed persistent bands of subsurface bubble clouds spaced by 5 to 10 m and extending in length up to 100 m. The clouds were aligned with the prevailing wind direction consistent with the organization expected from Langmuir circulation. Average downward velocities of 6 cm/s were observed in the bubble plumes that extended to a depth of 15 m. These early observations of near-surface processes motivated a series of instrumentation developments that have helped to explore the richness and complexity of movements within the ocean mixed layer.

11:20

2aAO6. Susy, Seascan, and other acoustical contraptions. Mark V. Trevorrow (Defence R&D Canada - Atlantic, 9 Grove St., PO Box 1012, Dartmouth, NS B2Y 3Z7, Canada, mark.trevorrow@drdc-rddc.gc.ca)

In the early 1990s, the acoustical oceanography research group, led by Dr. David Farmer, developed several high-frequency sonar platforms for near-surface oceanography. These platforms, suspended approximately 25 m below the surface, typically supported six frequencies of upward-looking echo-sounders and up to four steerable 100 kHz sidescan sonars. Additionally, the SUSY platform had four extensible arms, 4.5 m in length, each supporting a wideband hydrophone. The platforms had sufficient batteries and data recording for up to 40 hours of autonomous operation, which could be extended through scheduled on/off periods. The intent of these platforms was to investigate properties of near-surface bubble due to breaking waves, with additional Doppler velocity measurements of surface waves and convective circulations. These platforms were also used for other studies, such as tidal convergence zones and ship wakes. The SEACAN platform was also used to assess zooplankton and fish populations in lake in Japan. This presentation will review key field trials and scientific achievements generated through use of these platforms.

11:40

2aAO7. Passing on the excitement of experimental oceanography. Craig L. McNeil, Eric D'Asaro, and Andrey Shcherbina (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, cmcneil@apl.washington.edu)

Having graduated from the University of Victoria in 1995 with David Farmer as my advisor, I learned how exciting it is to work at the cutting-edge of experimental oceanography. During my Ph.D., I studied bubble mediated air-sea gas exchange which involved chasing winter storms in the NE Pacific by ship. That work inspired our study of the role of tropical cyclones in the global carbon cycle using air deployed gas sensing floats equipped with ambient noise recorders to evaluate wave breaking effects. I was also inspired when David enthusiastically showed me on the echosounder internal waves passing under the boat in Knight Inlet. I will present our recent AUV measurements made in prominent estuarine features (e.g., salt wedge and internal hydraulic jump) and discuss the impact of these features on suspended sediment distributions and underwater acoustic communications.

12:00

2aA08. Thirty years of scintillating acoustic data in diverse ocean environments, thanks to David Farmer. Daniela Di Iorio (Dept. of Marine Sci., Univ. of Georgia, 250 Marine Sci. Bldg., Athens, GA 30602, daniela@uga.edu)

The acoustic scintillation method was first applied in a coastal tidal channel in the early 1980s by David Farmer and this laid the foundation for studies of estuarine channel flows, bottom boundary layer dynamics, deep-sea hydrothermal plumes, and now more recently hydrocarbon seeps. Over short distances, using high frequencies and high transmission rates, amplitude and phase fluctuations measured over transmitter and receiver arrays have been used to infer horizontal (or vertical) flows and turbulent motions, all averaged along the acoustic path over the range separating transmitters and receivers. Autonomous and cabled instrumentation have provided measurements of temporally and spatially averaged quantities, continuously in time. This ability to make long-term continuous measurements has shown major advances in our understanding of acoustic forward scatter from velocity and temperature fluctuations in moving random media and for identifying strong turbulence levels in a variety of ocean settings. Much of the measurements described here would not have been possible without the generous contributions of David Farmer.

2a MON. AM

MONDAY MORNING, 26 JUNE 2017

ROOM 312, 9:15 A.M. TO 12:20 P.M.

Session 2aBAa

Biomedical Acoustics and Physical Acoustics: Impact of Soft Tissue Inhomogeneities and Bone/Air on Ultrasound Propagation in the Body

Vera Khokhlova, Cochair

University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Robin Cleveland, Cochair

Engineering Science, University of Oxford, Inst. Biomedical Engineering, Old Road Campus Research Building, Oxford OX3 7DQ, United Kingdom

Chair's Introduction—9:15

Invited Papers

9:20

2aBAa1. Fullwave simulations of ultrasound propagation in the human body: Applications to imaging and motion estimation. Gianmarco Pinton (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., 109 Mason Farm Rd., Taylor Hall Rm. 348, CB7575, Chapel Hill, NC 27599, gfp@unc.edu)

The Fullwave simulation tool, which solves a modified Westervelt equation, can model 3D ultrasound propagation in the human body. It includes the effects of nonlinearity, frequency dependent attenuation, and scattering. This high order finite difference simulation has a high dynamic range and ability to include sub-resolution scattering physics enabling it to tackle the computationally challenging problem of generating ultrasound images directly from the first principles of propagation and reflection. Three dimensional acoustical maps of the human body are derived from the Visible Human project, which provides the high degree of anatomical fidelity required to model sources of image degradation such as reverberation clutter and aberration. A three dimensional intercostal imaging scenario shows how the ribs degrade the image quality via reverberation and yet can improve the beam shape via apodization. It is shown that the in situ Mechanical Index (MI) in the human body differs significantly from derated estimates in a homogeneous medium. Due to tissue heterogeneities, the peak MI is close to the transducer and far from the focal region. Finally, we present a numerical model of subresolution displacements by using an impedance flow model applied to a scatterer composed of two elements. This shows that the numerical model can support displacements that are over three orders of magnitude smaller than the grid spacing. Applications to acoustic radiation force, elastography, and shear shock wave tracking are discussed.

9:40

2aBAa2. Effects of soft tissue inhomogeneities on nonlinear propagation and shock formation in high intensity focused ultrasound beams. Petr V. Yuldashev, Anastasia S. Bobina (Phys. Faculty, M.V. Lomonosov Moscow State Univ., 119991, Russian Federation, Moscow, Leninskie Gory, Moscow 119991, Russian Federation, petr@acs366.phys.msu.ru), Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), Adam D. Maxwell, Wayne Kreider (Appl. Phys. Lab., Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), George R. Schade (Dept. of Urology, Univ. of Washington, Seattle, WA), Oleg Sapozhnikov, and Vera Khokhlova (Phys. Faculty, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation)

Recent pre-clinical studies on boiling histotripsy (BH) of kidney *in vivo* have shown that the presence of soft tissue inhomogeneities such as skin, fat, and muscle layers do not prevent shock formation required for treatment. However, the increase in source power required to compensate for tissue aberrations may be impractically high and result in nearfield damage. Simulations can provide deeper insight into the impact of aberrations on shock formation in tissue and mitigation methods. A previously developed numerical solver of the Westervelt equation was refined to account for tissue inhomogeneities assuming that backward scattering effects can be neglected. Irradiation of human kidney using a single-element 1-MHz transducer with 5-cm radius and 9-cm focal distance was considered. Spatial distributions of the sound speed and mass density in tissue were reconstructed directly from a 3D CT scan. Values of nonlinear and absorption coefficients at each spatial location were assigned by CT image segmentation, using known literature data for different tissue types. Results are illustrated and discussed, including degradation of the focal maximum, shock formation, power compensation requirements, and the potential for modeling the impact of inhomogeneities through the use of an elevated equivalent attenuation. [Work supported by RSF-14-12-00974 and NIH R01EB7643.]

10:00

2aBAa3. Atlas-based simulations of high-intensity focused ultrasound. Bradley Treeby (Medical Phys. and Biomedical Eng., Univ. College London, Biomedical Ultrasound Group, Wolfson House, 2-10 Stephenson Way, London NW1 2HE, United Kingdom, b.treeby@ucl.ac.uk) and Jiri Jaros (Faculty of Information Technol., Brno Univ. of Technol., Brno, Czech Republic)

In silico investigations of high-intensity focused ultrasound (HIFU) have many applications, including patient selection (determining whether a patient is a good candidate for therapy), treatment verification (determining the cause of adverse events or therapy failures), and treatment planning (determining the optimum sonication parameters before therapy). Here, we use a patient atlas to study the effects of tissue heterogeneities on HIFU sonications in the kidney and liver. The patient atlas is derived from a segmentation of digital cryosection images from the Visible Human Project run by the U.S. National Library of Medicine. For each organ, simulations were repeated under both linear and nonlinear conditions, for different driving frequencies, and for several artificial configurations. These included using a constant sound speed, constant impedance, no absorption, and without particular anatomical features (e.g., the muscle, skin, and ribs). The relative importance of absorption, reflection, refraction, nonlinearity, and frequency is described. The parallel computing requirements needed to perform these large-scale full-wave ultrasound simulations in heterogeneous and absorbing media are also discussed.

10:20

2aBAa4. Thermodynamically viable wave equations for power law attenuation in viscoelastic media. Sverre Holm (Univ. of Oslo, Gaustadalleen 23B, Oslo N 0316 Oslo, Norway, sverre@ifi.uio.no)

Many complex media of great practical interest, such as in medical ultrasound, display an attenuation that increases with a power law as a function of frequency. Usually, measurements can only be taken over a limited frequency range, while wave equations often model attenuation over all frequencies. There is therefore some freedom in how the models behave outside of this limited interval, and many different wave equations have been proposed, in particular, fractional ones. In addition, it is desirable that a wave equation models physically viable media and for that two conditions have to be satisfied. The first is causality, and the second is a criterion that comes from thermodynamic considerations and implies that the relaxation modulus is a completely monotonic function. The latter implies that attenuation asymptotically cannot rise faster than frequency raised to the first power. These criteria will be explained and used to evaluate several of the ordinary and fractional wave equations that exist.

10:40

2aBAa5. Nonlinear ultrasound simulations using the discontinuous Galerkin method. James F. Kelly (Dept. of Statistics and Probability, Michigan State Univ., 1. East Lansing, MI 48824, kellyja8@stt.msu.edu), Xiaofeng Zhao, and Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Histotripsy with nonlinear ultrasound is an emerging noninvasive therapeutic modality that generates cavitation with high-intensity shock waves to precisely destroy diseased soft tissue. Numerical simulation of these shock waves in nonlinear, absorbing media is needed to characterize histotripsy systems and optimize treatments. We are developing a discontinuous Galerkin code based on the Westervelt equation to simulate transient wave propagation in the brain and skull. The discontinuous Galerkin method is a good choice for this simulation problem since this approach has high-order accuracy, geometric flexibility, low dispersion error, and excellent scalability on massively parallel machines. The Westervelt equation is formulated in a first-order flux form and discretized using a strong form of the discontinuous Galerkin method. Numerical results, both linear and nonlinear, from 1D and 2D discontinuous Galerkin codes, are presented and compared to both analytical and numerical benchmark solutions. In particular, the discontinuous Galerkin method captures nonlinear steepening of a high-intensity pulse with minimal numerical artifacts. The development of a 3D massively parallel code is also briefly discussed. [This work was supported in part by a grant from the Focused Ultrasound Foundation.]

11:00

2aBAa6. Focusing ultrasound through bones: Past, current, and future transcostal and transskull strategies. Jean-Francois Aubry (Institut Langevin, CNRS, 17 rue Moreau, Paris 75012, France, jean-francois.aubry@espci.fr)

Bones reflect, refract, distort, and absorb ultrasonic waves. Most medical application of ultrasound avoid bony structures. Nevertheless, for liver and brain therapy, the rib cage and the skull are in the ultrasonic path. We will present non-invasive methods to detect the presence of the ribs and shape the beam in order to sonicate in between the ribs. It is thus possible to non-invasively aim a target shadowed by the rib cage while avoiding sonication through the bones. Such approaches take advantage of the ultrasonic imaging capabilities of multi-element arrays. Nevertheless, non-invasive brain therapy requires to sonicate through the skull. CT and MR- based techniques have been developed to estimate the phase shifts induced by the skull. Dedicated multi-element arrays have been developed to generate the appropriate phase-corrected signals. The number of elements of the arrays has progressively increased in order to shape at best the corrected beam (64 elements in 2000, 300 in 2003, and 1024 in 2012): clinical transcranial therapies will be presented for the treatment of essential tremor, parkinsonian tremors, and tumors with a 1024 element array. We will also present preliminary results of a novel game changing transcranial focusing technique requiring a dramatically lower number of elements.

Contributed Papers

11:20

2aBAa7. Focusing ultrasound through the skull for neuromodulation.

Joseph Blackmore (Inst. of Biomedical Eng., Univ. of Oxford, University of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, joseph.blackmore@wadh.ox.ac.uk), Michele Veldsman, Christopher Butler (Nuffield Dept. of Clinical NeuroSci., Univ. of Oxford, Oxford, United Kingdom), and Robin Cleveland (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Focused ultrasound for neuromodulation is emerging as a non-invasive brain stimulation method, whereby low-intensity pulsed ultrasound is focused through the skull to locations within the brain. The ultrasound results in excitation of the targeted brain region, and stimulation in the motor and visual centers has already been reported. One barrier is that the strongly heterogeneous skull bone distorts, aberrates, and attenuates the ultrasound beam leading to disruption and shifting of the focus. While transducer arrays can be used to correct for these aberrations, this equipment is expensive and complex. Here, numerical modeling is used to determine the optimal placement of a single element focused transducer to achieve the required focusing. Numerical simulations, using a point source at target locations in the visual cortex, are employed to determine the phase and amplitude on a spherical surface placed outside the head. The optimal placement of the transducer is determined by minimizing the weighted phase error over the transducer surface. Appropriate focusing is then confirmed by simulating the pressure field in the brain tissue for the optimal transducer location. Both elastic and fluid-type models of the skull are considered to assess the impact of shear waves on the targeting.

11:40

2aBAa8. Imaging cortical bone using the level-set method to regularize travel-time and full waveform tomography techniques. Jonathan R. Fincke (Mech. Eng., Massachusetts Inst. of Technol., 99 Hancock St., Apt. 10, Cambridge, MA 01239, jfincke@mit.edu)

Regularizing travel-time and full waveform tomographic techniques with level set methods enables the recovery of cortical bone geometry. Bone

imaging and quantification is of general clinical utility. Applications include prosthetic fitting, fracture detection, and the diagnosis of osteoporosis as well as monitoring the disease's progression. Recently, there is significant interest in imaging through the skull, imaging long bones as well as estimating their material properties. In this case, a cylindrical acquisition geometry is used, allowing the bone to be imaged from all angles. Frequencies between 200 kHz and 800 kHz are used to enable penetration into the bone region.

12:00

2aBAa9. Flash focus ultrasonic images sequences for shear shock wave observation in the brain. David Espindola and Gianmarco Pinton (Dept. of Biomedical Eng., Univ. of North Carolina at Chapel Hill, 109 Mason Farm Rd., Taylor Hall Rm. 348, Chapel Hill, NC 27599, daanesro1@gmail.com)

Nonlinear shear waves have a cubic nonlinearity which results in the generation of a unique characteristic odd harmonic signature. This behavior was first observed in a homogeneous gelatin phantom using ultrafast plane wave ultrasound imaging and a correlation-based tracking algorithm to determine particle motion. However, in heterogeneous tissue, like brain, the heterogeneities generate clutter that degrades motion tracking to the point where the shock waves and their characteristic odd harmonics are no longer observable. We present a high frame-rate ultrasound imaging sequence consisting of multiple focused emissions that improves the image quality and reduces clutter to generate high quality motion estimates of shear shock waves propagating in the brain. A point spread function analysis is used to characterize the improvements of the proposed imaging sequence. It is shown that the flash focus sequence reduces the side lobes by 20 dB while retaining the same spatial resolution translating to a sensitivity up to the 11th harmonic. The flash focus sequence are then used to acquire high frame-rate (6500 fps) ultrasound movies of an ex-vivo porcine brain in which a shear wave propagates. Using an adaptive tracking algorithm, we compute the particle velocity in a field of view as deep as the brain. It is therefore demonstrated that the proposed method can detect the nonlinear elastic motion and the odd harmonics with sufficient sensitivity to observe the development of a shear wave into a shock wave as it propagates in the brain.

2a MON. AM

Session 2aBAb

Biomedical Acoustics: Beamforming and Image Guided Therapy III: Ablation and Histotripsy

Costas Arvanitis, Cochair

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

Constantin Coussios, Cochair

*Department of Engineering Science, Institute of Biomedical Engineering, University of Oxford, Old Road Campus Research Building, Oxford OX3 7DQ, United Kingdom**Invited Paper*

9:20

2aBAb1. Real-time ablation monitoring and lesion quantification using harmonic motion imaging guided focused ultrasound (HMIgFUS). Elisa Konofagou (Columbia Univ., 1210 Amsterdam Ave., ET351, New York, NY 10027, ek2191@columbia.edu)

High-Intensity Focused Ultrasound (HIFU) monitoring is currently hindered by time- and cost-inefficient or inconclusive warranting, thus an imaging technique for efficient and reliable guidance. Harmonic Motion Imaging (HMI) uses a focused ultrasound (FUS) beam to generate an oscillatory acoustic radiation force for an internal, non-contact palpation to internally estimate relative tissue hardness. HMI also uses ultrasound imaging with parallel beamforming and estimates and maps the tissue dynamic motion in response to the oscillatory force at the same frequency based on consecutive RF frames. HMI has already been shown feasible in simulations, phantoms, *ex vivo* human and bovine tissues as well as animal tumor models *in vivo*. Using an FUS beam, HMI can also be seamlessly integrated with thermal ablation using HIFU, which leads to changes in the tumor stiffness. In this paper, an overview of HMI will be provided, including the capability of HMI to characterize and image the tumor prior to ablation, localize the beam for treatment planning, as well as monitor subsequent lesioning in real time. The findings demonstrate that HMI is capable of both detecting and characterizing the tumor prior to HIFU ablation as well as correctly depict and quantify the lesion during treatment. More importantly, HMI is shown capable of distinguishing the tumor margins from those of the thermal lesion *in vivo* in order to accurately determine treatment success. HMI thus constitutes an integrated, real-time method for efficient HIFU monitoring.

Contributed Papers

9:40

2aBAb2. Real-time feedback control of high-intensity focused ultrasound thermal ablation using echo decorrelation imaging. Mohamed A. Abbass, Jakob K. Killin, Neeraja Mahalingam, and T. Douglas Mast (Biomedical Eng. Program, Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, abbassma@mail.uc.edu)

The feasibility of controlling high-intensity focused ultrasound (HIFU) thermal ablation in real time using echo decorrelation imaging feedback was investigated in *ex vivo* bovine liver. Sonication cycles (5.0 MHz, 0.7 s per HIFU pulse, 20-24% duty, 879-1426 W/cm² spatial-peak, and temporal peak intensity) performed by a linear image-ablate array were repeated until the minimum cumulative echo decorrelation within the focal region of interest exceeded a predefined threshold. Based on preliminary experiments ($N=13$), a threshold of -2.7 for the \log_{10} -scaled echo decorrelation per millisecond was defined, corresponding to 90% specificity of local ablation prediction. Controlled HIFU thermal ablation experiments ($N = 10$) were compared with uncontrolled experiments employing 2, 5, or 9 sonication cycles. Controlled trials showed significantly smaller average lesion area (4.78 mm²), lesion width (1.29 mm), and treatment time (5.8 s) than 5-cycle (7.02 mm², 1.89 mm, 14.5 s) or 9-cycle (9.31 mm², 2.4 mm, 26.1 s) uncontrolled trials. Prediction of local ablation using echo decorrelation was assessed using receiver operator characteristic (ROC) curve analysis, in which controlled trials showed significantly greater prediction capability (area under the ROC curve AUC=0.956) compared to 2-cycle uncontrolled

trials (AUC=0.722). These results suggest that ablation control using echo decorrelation may improve the precision, reliability, and duration of ultrasound-guided HIFU treatments.

10:00

2aBAb3. Sub-millimeter bistatic passive acoustic mapping. Delphine Elbes, Catherine Paverd, Robin Cleveland, and Constantin Coussios (Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg., Headington, Oxford OX3 7DQ, United Kingdom, delphine.elbes@eng.ox.ac.uk)

Passive acoustic mapping (PAM) is an emerging technique used to image sources of non-linear acoustic emissions, such as inertially cavitating bubbles, during ultrasound therapy. When using a conventional diagnostic ultrasound array, the transverse resolution is typically an order of magnitude better than the axial resolution, which may be inadequate for monitoring treatment at acoustically large distances from the array. Here, we describe an experimental technique that utilizes two orthogonal and coplanar 128-element linear arrays (6.25 MHz centre frequency) to overcome this limitation. The resolution of bistatic PAM was characterized by varying the distance between sources, as well as the array-to-sources distance. The optimal number of elements required was identified by considering the resolution, the accuracy in quantifying the energy of acoustic emissions, and the computational cost. The resulting resolution (achieved with 256 elements) was close to theoretical transverse resolution limit, on the order of hundreds of microns, and the advantage of bistatic PAM over conventional

monostatic PAM was illustrated in the context of non-invasive fractionation of the intervertebral disc. It is concluded that, at depth, bistatic PAM enables improved real-time treatment monitoring on biologically relevant length scales. [Work supported by UK Engineering and Physical Sciences Research Council (EP/K020757/1).]

10:20

2aBAb4. Real-time acoustic-based feedback for histotripsy therapy. Jonathan J. Macoskey, Jonathan R. Sukovich, Timothy L. Hall, Charles A. Cain, and Zhen Xu (Biomedical Eng., Univ. of Michigan, Carl A. Gerstacker Bldg., 2200 Bonisteel Blvd., Ann Arbor, MI 48109, macoskey@umich.edu)

Histotripsy uses high-pressure microsecond ultrasound pulses to generate cavitation to fractionate cells in target tissues. Two acoustic-based feedback mechanisms are being investigated to monitor histotripsy therapy in real-time. First, bubble-induced color Doppler (BICD) is received by an ultrasound probe co-aligned with the histotripsy transducer to monitor the cavitation-induced motion of residual cavitation nuclei in tissue throughout treatment. Second, acoustic backscatter of the histotripsy pulse from the cavitation bubbles is received by directly probing elements of histotripsy transducer to monitor acoustic emissions from the cavitation bubbles during treatment. In these experiments, histotripsy was applied to agarose phantoms and *ex vivo* tissue by a 112-element, 500 kHz semi-hemispherical ultrasound array with a 15 cm focal distance. The BICD signals were collected on a Verasonics system by an L7-4 probe. The BICD and backscatter signals were compared to high-speed optical images of cavitation in phantoms and histology of tissue. A consistent trend was observed in both the BICD and backscatter waveforms throughout treatment in both tissue and agarose phantoms that correlated with high-speed imaging and histological analysis. These results suggest that BICD and acoustic backscatter can provide non-invasive, real-time, quantitative feedback of tissue treatment progression during histotripsy, thus improving treatment efficiency and accuracy.

10:40–11:00 Break

11:00

2aBAb5. Characterization of cavitation-radiated acoustic power using single-element detectors. Kyle T. Rich and T. Douglas Mast (Biomedical Eng. Program, Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267, richkylet@gmail.com)

Standard approaches to quantifying cavitation activity using emission measurements made by single-element passive cavitation detectors (PCD) would be facilitated by improved quantitative and system-independent characterization techniques. Although the strength of an individual emission source can be determined from absolute pressure measurements by a calibrated PCD, this approach requires spatially resolved detection of single bubbles at known locations. Here, a method is shown for characterizing an ensemble of emission sources, quantified by their radiated acoustic power per unit area or volume of a defined region of interest (ROI). An analytic diffraction-correction factor relating frequency-dependent PCD-measured pressures to cavitation-radiated acoustic power is derived using a spatial integral of the PCD sensitivity. This approach can be applied to measurements made by any PCD without *a priori* knowledge of the number or spatiotemporal distribution of cavitation bubbles. Simulations show that cavitation-radiated acoustic power per unit ROI volume or area is accurately recovered by compensation of emissions received by focused or unfocused PCDs. Measurements from previous sonophoresis experiments are analyzed, showing that skin permeability changes from 0.41 or 2.0 MHz sonication are comparably correlated to the radiated acoustic power of subharmonic emissions per unit area.

11:20

2aBAb6. Acoustic radiation force on a sphere in tissue due to the irrotational component of the shear field body force. Benjamin C. Treweek, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, btweek@utexas.edu)

Acoustic radiation force on a sphere in soft tissue can be written as the sum of four distinct contributions. Two arise from incident and scattered compressional waves only, one from direct integration of the time-averaged Piola-Kirchhoff stress tensor over the surface of the sphere, and one from the irrotational component of the body force producing deformation of the surrounding medium. The other two contributions also incorporate scattered shear waves, and they are found by the same procedures. Three of these terms are known analytically [Ilinskii *et al.*, Proc. Meet. Acoust. **19**, 045004 (2013)], but the contribution relating to the shear field body force must be found numerically. Preliminary results for this term were obtained through simplifying approximations and presented at the fall 2016 ASA meeting; the present submission extends this work to cases where these approximations do not hold. Helmholtz decomposition of the shear field body force is performed using 3D Fourier transforms, then the irrotational potential is integrated over the surface of the sphere. Various sphere materials are considered, and comparisons are made with known results for a sphere in an ideal fluid. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

11:40

2aBAb7. Investigation of the source of histotripsy acoustic backscatter signals. Jonathan R. Sukovich, Timothy L. Hall, Jonathan J. Macoskey, Charles A. Cain, and Zhen Xu (Biomedical Eng., Univ. of Michigan, 1410 Traver Rd., Ann Arbor, MI 48105, jsukes@umich.edu)

Recent work has demonstrated that acoustic backscatter signals from histotripsy-generated bubble clouds may be used to localize generated bubble clouds and perform non-invasive aberration correction transcranially. However, the primary source of the measured signals, whether from emissions generated during bubble expansion, or scattering of the incoming pulses off of the incipient bubble clouds, remains to be determined and may have important implications for how the acquired signals may be used. Here, we present results from experiments comparing the acoustic emissions and growth-collapse curves of single bubbles generated optically to those generated via histotripsy. Histotripsy bubbles were generated using a 32-element, 1.5 MHz spherical transducer with pulse durations < 2 -cycles; optical bubbles were nucleated using a pulsed Nd:YAG laser focused at the center of the histotripsy transducer. Optical imaging was used to capture the time evolution of the generated bubbles from inception to collapse. Acoustic emissions from the generated bubbles were captured using the receive-capable histotripsy transducer elements as well as with a commercial hydrophone mounted within. Imaging results indicated that optically nucleated bubbles experienced more rapid growth than histotripsy generated bubbles. Acoustic emissions from both sets of bubbles were comparable, however, suggesting the primary component of the measured histotripsy “backscatter” signal is an emission generated during bubble expansion.

12:00

2aBAb8. Cylindrically converging nonlinear shear waves. John M. Cormack, Kyle S. Spratt, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

The low shear moduli of soft elastic media permit the generation of shear waves with large acoustic Mach numbers that can exhibit waveform distortion and even shock formation over short distances. Waves that converge onto a cylindrical focus experience significant dispersion, causing waveforms at the focus and in the post-focal region to differ significantly from the source waveform even in the absence of nonlinear distortion. A full-wave model for nonlinear shear waves in cylindrical coordinates that accounts for both quadratic and cubic nonlinearity is developed from first principles. For the special case of an infinite cylindrical source with particle motion parallel to the axis, for which nonlinearity is purely cubic, the nonlinear wave equation is solved numerically with a finite-difference scheme.

The full-wave model is compared with a piecewise model based on a generalized Burgers equation for cylindrically converging waves outside of the focal region and linear diffraction theory in the focal region. For waveforms with wavelength much smaller than the source radius, conditions are

explored for which the approximate piecewise model shows good agreement with the full-wave model. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

MONDAY MORNING, 26 JUNE 2017

ROOM 205, 9:15 A.M. TO 12:20 P.M.

Session 2aEA

Engineering Acoustics: Ducts and Mufflers I

Mats Åbom, Cochair

The Marcus Wallenberg Laboratory, KTH-The Royal Inst. of Technology, Teknikringen 8, Stockholm 10044, Sweden

David Herrin, Cochair

Department of Mechanical Engineering, University of Kentucky, 151 Ralph G. Anderson Building, Lexington, KY 40506-0503

Chair's Introduction—9:15

Invited Papers

9:20

2aEA1. Design, development, and implementation of low cost high performance mufflers for heavy duty diesel engines. Mohan D. Rao (Mech. Eng., Tennessee Tech, Box 5014, Box 5014, Cookeville, TN 38505, mrao@tntech.edu)

In this paper, details on the design and fabrication of affordable high-performing passive exhaust mufflers and associated low volume manufacturing technology for commercial heavy duty diesel engines are presented. The exhaust noise radiation to the atmosphere from large diesel engines used in earth-moving, military, and other heavy equipment machines ranks as a major noise source of the urban environment. A solution to this problem is the use of mufflers to significantly reduce noise pollution to the atmosphere. There are several types of common mufflers, including reactive designs using resonators, expansion chambers, perforations, and dissipative configurations using absorptive materials. The design and realization of mufflers for a particular engine is dependent on several design parameters, such as internal combustion engine characteristics, acoustical requirements by standards, production volumes and cost, with the requirements varying from application to application. Particulars of successful muffler designs for three different machines—a commercial excavator, a tele handler, and a military ground vehicle are presented in this paper. The designs include both box type and round mufflers with Helmholtz resonators, reversible flow chambers, and perforated tubes, all capable of custom fabrication using conventional manufacturing processes in a local machine shop using commercially available materials.

9:40

2aEA2. On challenges and considerations in designing cold end exhaust systems for current and future automotive applications. Raghavan Vasudevan (Exhaust R&D, Magneti Marelli, 3900 Automation Ave., Auburn Hills, MI 48326, raghavan.vasudevan@magnetimarelli.com)

Efforts to meet ever-stringent fuel economy standards has led to increased focus on light weighting technologies, and proliferation of alternative powertrains such as hybrid vehicles. This has significantly increased the complexity of designing exhaust systems to meet high level NVH performances against thin powertrain targets. In this paper, some applications will be presented with latest efforts in addressing some of these challenges. (a) Novel methodologies in predicting exhaust high frequency flow noise using transient CFD simulations will be discussed. (b) Customized tuning applications: Case study in optimization, Daisy-chaining, and positioning of narrow frequency tuning applications (e.g., concentric tube Helmholtz resonators with tuned orifices, pipe length tuning) driven by tighter clearances and directed light weighting efforts will be discussed. (c) Hybrid applications—Hybrid powertrains pose a unique challenge as the engine can function both as power plant and a power generator, both of which have different NVH requirements. Efforts in addressing these challenges with limited packaging space will be discussed.

10:00

2aEA3. Application of micro-perforate tubes in motorcycle mufflers. Henry C. Howell (ProDC Development Ctr., Harley-Davidson Motor Co., ProDC Development Ctr., 11800 W. Capitol Dr., Wauwatosa, WI 53222, hank.howell@harley-davidson.com)

Regulator noise requirements, torque and horsepower goals, and sound quality targets drive the acoustic performance expected from motorcycle mufflers, but exposed motorcycle mufflers also become important styling features on bikes. Weight, shape, temperature, rider leg position, coatings, and cosmetics all become factors which influence the mufflers final design. With these factors in mind, the uses of micro-perforated metals in grazing flow applications have allowed improvements in all of these areas.

10:20

2aEA4. Sensitivity study of exhaust system using the Moebius transformation. Yitian Zhang (W. L. Gore and Assoc., 1025 Christina Mill Dr., Newark, DE 19711, yitzhang@wlgore.com)

The performance of exhaust system is not only dependent on the system itself, but also on the boundary conditions, which are the impedances at the inlet and outlet. For many cases, the exact value of these impedances are not known or easily measured. It is of interest to see the range of performance variation, given the range of possible values of impedance. An exhaustive method to determine the response variation can be used, but is computationally expensive. However, it can be proved that the relationship between boundary impedance and response is in the form of the Moebius transformation, which is a conformal transformation. Taking advantage of this property, the computation can be much reduced. It is also shown that the sensitivity of this dependence can be studied visually using the Moebius transformation.

10:40

2aEA5. Acoustic performance of an annular Helmholtz resonator and its application in exhaust system. Xin Hua and Jagdish Dholaria (Faurecia Emissions Control Technologies, 950 West 450 South, Columbus, IN 47201, xin.hua@faurecia.com)

Helmholtz resonator is a traditional acoustic tuning component, which consists of an enclosed volume and a throat neck. It is tuning and packaging friendly and thus has been widely used in exhaust systems. An annular Helmholtz bottle resonator usually has a straight-through pipe with a certain length of perforation. A larger sleeve is mounted to cover the straight-through pipe with one end open and the other end closed. The annular volume gap between the straight-through pipe and the sleeve becomes the neck of the Helmholtz resonator. In this research, an annular Helmholtz resonator is investigated. Numerical simulation and non-linear least square method are used to propose a correction on empirical equation to estimate the resonator targeting frequency. Afterward, this type of Helmholtz resonator is applied to a two-box exhaust system. Insertion loss is used to investigate the resonator performance in the system.

11:00

2aEA6. Noise suppressors with engineered compliance in fluid hydraulic systems. Kenneth Cunefare, Elliott Gruber, and Nathaniel Pedigo (Georgia Tech, Mech. Eng., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

Fluid power hydraulic systems, common on a wide variety of industrial and construction equipment, frequently exhibit undesirable noise characteristics. The noise is primarily due to the pump-induced pressure pulsation in the fluid. One means to control this fluid-borne noise is through the use of a suppressor integrating compliance, as may be introduced using a pressurized bladder, or an elastic compliant liner exposed to the fluid. This compliance causes an impedance change at the inlet to the suppressor. Within the suppressor, the compliance leads to a reduced sound speed, which may then lead to fluid particle velocities high enough for damping to become effective. Classical dissipative noise control means, as is common in air or gas mufflers, is otherwise ineffective in fluid systems because of the low particle velocity. An engineered solid material, a syntactic foam, is under development for use in hydraulic systems. The syntactic foam of interest is comprised of microspheres dispersed in a polymer. Particular challenges include retaining functionality at high system pressures, which may be addressed by pressurizing the microspheres. The foam's performance increases with volume fraction of microspheres, and internal pressurization. The material is enabling of a other fluid noise control devices, including water hammer arrestors.

11:20

2aEA7. Proper integration of plane wave models into the design process. David Herrin (Dept. of Mech. Eng., Univ. of Kentucky, 151 Ralph G. Anderson Bldg., Lexington, KY 40506-0503, dherrin@enr.uky.edu) and Tamer Elnady (Ain Shams Univ., Cairo, Egypt)

Muffler and silencer design is primarily accomplished through cut and try approaches in many industries. Although test mufflers are inexpensive to manufacture and test, better designs can be arrived at rapidly and less expensively using plane wave methodologies. Moreover, engineers develop intuition. The current work is aimed at demonstrating how plane wave models can be integrated into the design process. It is first demonstrated that plane wave models can reliably determine the performance of complicated mufflers below the cutoff frequency. Tips for developing plane wave models are summarized.

11:40

2aEA8. Sound quality control of axial fan noise by using microperforated panel housing with a hollow tube. Yatsze Choy, Yan Kei Chiang, and Qiang Xi (FG639 Dept. of ME, The Hong Kong Polytechnic Univ., Hong Kong 852, Hong Kong, mmyschoy@polyu.edu.hk)

This study presents a novel passive noise control approach to directly suppress sound radiation from an axial-flow fan, which involves micro-perforated panels (MPP) backed by cavities and a hollow tube. Apart from the sound suppression performance in terms of insertion loss, sound quality of axial fan with a dipole nature is also investigated which serves as a significant supplementary index for assessing the noise control device. The noise suppression is achieved by the sound cancelation between sound fields from the fan of dipole nature and sound radiation from a vibrating panel via vibro-acoustic coupling and interference from the hollow tube boundaries,

as well as by sound absorption in micro-perforations. A two-dimensional theoretical model, capable of dealing with strong coupling among the vibrating micro-perforated panel, sound radiation from the dipole source, and sound fields inside the cavity and the duct is developed. The theoretical results are validated by both finite element simulation and experiment. Results show that an addition of hollow tube enhances the sound suppression performance in the passband region of the MPP housing device. The findings of the current research have the potential to control ducted-fan noise effectively, to enhance the quality of products with a ducted-fan system.

12:00

2aEA9. Revisiting the Cremer impedance. Raimo Kabral, Mats Åbom (The Marcus Wallenberg Lab., KTH-The Royal Inst. of Technol., KTH-The Marcus Wallenberg Lab., Stockholm 10044, Sweden, kabral@kth.se), and Börje Nilsson (Dept. of Mathematics, Linnaeus Univ., Växjö, Sweden)

In a classical paper (*Acustica* 3, 1953), Cremer demonstrated that in a rectangular duct, with locally reacting walls, there exists an impedance (“the Cremer impedance”) that maximizes the propagational damping for the lowest mode. Later (*JSV* 28, 1973), Tester extended the analysis to include a plug flow and ducts of both circular and rectangular cross-section. One limitation in the work of Tester is that it simplified the analysis of the effect of flow only considering high frequencies or well cut-on modes. This approximation is reasonable for large duct applications, e.g., aeroengines, but not for many other cases of interest. Kabral *et al.* (*Acta Acustica united with Acustica* 102, 2016) removed this limitation and investigated the exact Cremer impedance including flow effects. As demonstrated in that paper the exact solution exhibits some special properties at low frequencies, e.g., a negative real part of the wall impedance. In this paper, the exact Cremer impedance is further analyzed and discussed.

MONDAY MORNING, 26 JUNE 2017

ROOM 304, 9:35 A.M. TO 12:20 P.M.

Session 2aEDa

Education in Acoustics, Public Relations Committee, and Student Council: Communicating Scientific Research to Non-Scientists

Andrew A. Piacsek, Cochair

Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926

Kerri Seger, Cochair

Scripps Institution of Oceanography, 9331 Discovery Way, Apt. C, La Jolla, CA 92037

Chair’s Introduction—9:35

Invited Papers

9:40

2aEDa1. Shouting across the chasm. Soren Wheeler (Radiolab, New York Public Radio, 25 N 6th St., Madison, WI 53704, swheeler@wnyc.org)

As people are interested in science, we assume science is interesting. But if you want to be an effective science communicator, you must assume the exact opposite. After more than 20 years working on the public understanding of science, the thing that continues to astonish and trouble me is how many people shudder at the word “science,” or at any sniff of the words, cadences, or even attitudes associated with science. And yet those are the people we most desperately need to reach. Thus, to communicate science is to cross a chasm of interest, knowledge, inclination, and language. With a few examples, I will talk about the ways that voice, structure, and emotional experience can help us shout across that chasm, and the ways in which audio storytelling, in particular, can shine a light on how to get people to listen to us when we talk about science.

10:00

2aEDa2. Talking science with non-scientists. Karin Heineman (Executive Producer, Inside Sci., American Inst. of Phys., One Phys. Ellipse, College Park, MD 20740, kheinema@aip.org)

Can you explain your research to a non-scientist, or even a 7th grader in 90 seconds or less—and explain it in a way that’s engaging and understandable? Communicating science to an audience of non-scientists requires a very different skill set than giving a successful talk to a conference or exchanging ideas with peers or colleagues. Effective science communication skills are important to have—you’re

providing lay audiences with accurate and reliable science information, while protecting the public from misinformation, and also influencing public policy and inspiring a sense of wonder in non-scientists. This talk is targeted for a broad audience of scientists who would like to learn how science communication skills will improve their ability to promote their research and get more media attention. A huge communication tool is video. Don't be afraid to be on camera—learn what to expect from an on-camera interview; learn tricks to get your message across; learn what impact the right visuals can have in communicating your message and research. Learn the tools and tricks of the trade of communicating science from speech, to tone, to what not to wear— and make a lasting impression on any audience.

10:20

2aEDa3. Communicating science 101. Brad Lisle (Foxfire Interactive, 500 East Washington St., Ste. 30, North Attleboro, MA 02760, brad@foxfireinteractive.com)

What's the best way to communicate science to non-scientists? Learn how producer/director Brad Lisle tackles this question through two different science media projects. The first, ZOOM into Engineering, targets engineers who are interested in doing hands-on engineering activities with K-6 students. The second, The Global Soundscapes Project, targets middle school students and the general public and explores the emerging field of soundscape ecology.

10:40–11:00 Break

11:00

2aEDa4. Joe McMaster: “Telling your story and bringing abstract ideas to life on screen.” Joseph McMaster (Sci. Journalist & Filmmaker, 700 Technol. Sq., Cambridge, MA 02139, joemcmaster1@gmail.com)

Mark Twain said, “I like a good story well told. That is the reason I am sometimes forced to tell them myself.” So how might you tell the story of your scientific research in engaging ways? And what can you do if non-scientists find the subject difficult or hard to visualize? Award winning science journalist and filmmaker Joe McMaster shares concrete examples of what has worked (and sometimes not worked) in his 25 years of making films about science and technology for television and the web—many of which have required bringing abstract ideas to life on screen.

11:20

2aEDa5. Integrating science communication into undergraduate and graduate curricula. Laura Kloepper (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, lkloepper@saintmarys.edu)

Effective science communication (“SciComm”) to both a technical and non-technical audience is considered a key skill for contemporary scientists. Unfortunately, most of the SciComm education occurs after individuals receive their degree, and few science students receive formal training in communication. In this presentation I will describe a course that teaches SciComm to first-year undergraduate science students and can be easily modified for graduate curricula. This course consists of a series of modules specific to various communication methods (oral presentations, formal writing, informal writing, and television/radio) and audiences. Students are exposed to SciComm through group assignments, written assignments, community experiences, and virtual lectures. I will describe these modules and give example assignments and assessment. By integrating SciComm early into science education, students can develop both the technical and communication skills necessary to be effective researchers and communicators.

11:40

2aEDa6. Don't cite it, write it. Raising awareness of acoustics through Wikipedia. Thais C. Morata (Div. of Appl. Res. and Technol., National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., M.S. C27, Cincinnati, OH 45226, tmorata@cdc.gov), Max Lum (Office of the Director, National Inst. for Occupational Safety and Health, Cincinnati, OH), James Hare (Office of the Director, National Inst. for Occupational Safety and Health, Washington, DC), and Leonardo Fuks (Escola de Musica, Universidade do Rio de Janeiro, Rio de Janeiro, Rio de Janeiro, Brazil)

Wikipedia is accessed by hundreds of millions around the world and that makes Wikipedia one of the most powerful platforms for the dissemination of science information. While Wikipedia offers high-quality content about certain topics, a large proportion of articles are insufficiently developed. The Wikimedia Foundation has engaged in partnerships with scientific and academic institutions to improve the coverage and communication of science to the public. These efforts are beneficial to professional and academic associations interested in sharing reliable, vetted information about their discipline with the world. The National Institute for Occupational Safety and Health (NIOSH) is one of the agencies engaged in this effort. NIOSH developed and manages the WikiProject Occupational Safety and Health. NIOSH also participated in a classroom program (where students write Wikipedia articles) to expand and improve Wikipedia's content on acoustics, noise, and hearing loss prevention. Faculty and students from the University of Rio de Janeiro contributed content on basic principles of acoustics. Metrics on these efforts are publicly available so reach can be evaluated by the number of views and quality of entries. Throughout these initiatives, new scientific content related to acoustics was successfully added to Wikipedia, and the quality of the entries were improved.

12:00–12:20 Panel Discussion

Session 2aEDb

Education in Acoustics: Education in Acoustics Poster Session

Eoin A. King, Chair

Mechanical Engineering, University of Hartford, 200 Bloomfield Avenue, West Hartford, CT 06117

All posters will be on display from 10:20 a.m. to 11:40 a.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 10:20 a.m. to 11:00 a.m. and authors of even-numbered papers will be at their posters from 11:00 a.m. to 11:40 a.m.

Contributed Papers

2aEDb1. A cross-university massive open online course on communication acoustics. Sebastian Möller (Quality and Usability Lab, TU Berlin, Sekr. TEL-18, Ernst-Reuter-Platz 7, Berlin 10587, Germany, sebastian.moeller@tu-berlin.de), Jens Ahrens (Audio Technol., Chalmers Univ. of Technol., Gothenburg, Sweden), M. Ercan Altinsoy (Inst. of Acoust. and Speech Commun., TU Dresden, Dresden, Germany), Martin Buchschmid (Chair for Structural Mech., TU München, München, Germany), Janina Fels (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany), Stefan Hillmann, Christoph Hohnerlein (Quality and Usability Lab, TU Berlin, Berlin, Germany), Gerhard Müller (Chair for Structural Mech., TU München, München, Germany), Bernhard U. Seeber (Audio Information Processing, TU München, Munich, Germany), Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany), Stefan Weinzierl (Fachgebiet Audiokommunikation, TU Berlin, Berlin, Germany), Sebastian Knoth, and Wolfram Barodte (Serviceeinheit Medien für die Lehre, RWTH Aachen, Aachen, Germany)

Four of the nine big Technical Universities in Germany, together with Chalmers University of Technology in Sweden, have developed a new Massive Open Online Course (MOOC) on the subject of Communication Acoustics. The idea is to foster education on the late Bachelor or early Master level by joining the expertise available at individual universities and by creating an online course offered both to local as well as remote students. The course started in winter term 2016 and is hosted on the EdX platform. It is offered in English language and roughly divided into two parts: The first part covers basics on acoustics, signal processing, human hearing, speech production, as well as electroacoustics and psychoacoustics. The second part introduces selected applications, such as sound recording and reproduction, sound fields and room acoustics, binaural technology, speech technology, as well as product sound design. The course material consists of explanatory videos and text as well as audiovisual material, exercises, and self-assessments. The final examination takes place as a written or online exam, with physical presence at the contributing sites. The talk will provide insights into the experiences we made, and illustrates how we overcame the obstacles inherent to cross-university education.

2aEDb2. Acoustics in African Music Instrument Technology: Training the Baton Bearer for Sustenance in Music Education in Nigeria. Stephen G. Onwubiko (Music, Univ. of Nigeria, Nsukka Enugu State, Enugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com)

With sparsely programs in musical acoustics located and taught throughout the country's institutions, the role of teaching music acoustics in higher education has received no attention with less trained teachers/educators, which has also become a controversial issue among scholars exploring that aspect of education. This paper shows ideological-opinions on integrating acoustics trained educators in African music instrument technology into the curriculum, since its technology invariably stress on acoustics which is an integral part of music technology and African music instrument technology alike. *Teachers possess the power to create conditions that can help*

students learn a great deal—or keep them from learning much at all. Teaching is an intentional act of creating those conditions according to Parker Palmer, (1998). The technology of any musical instrument is to satisfy the ear and heart acoustically speaking, for each musical instrument ranging from western instrument to the traditional African instrument has its technology, its sound, pitch, frequency and vibrations likewise these instrument vary in pitch, sound, frequency, tonal reflection, texture and color. This paper constantly explores these issues through qualitative interviews and observations with teachers and scholars on training the baton bearer in African musical acoustics for sustenance in music education in Nigeria.

2aEDb3. Self-taught Finite Element Method implementation to the assessing of natural frequencies and modal shapes in 2D rectangular plates. Augusto R. Carvalho de Sousa (Lab. of Vibrations and Acoust., Dept. of Mech. Eng., Federal Univ. of Santa Catarina, Laboratório de Vibrações e Acústica, Universidade Federal de Santa Catarina, Florianópolis, Santa Catarina 88040-900, Brazil, augusto_carvalho@live.com) and Jeferson R. Bueno (Civil Eng. Academic Dept., Federal Technology Univ. of Paraná – UTFPR, Campo Mourão, Paraná, Brazil)

This work presents a self-taught guide to the implementation of a Finite Element Model (FEM) to assess the natural frequencies and modal shapes of 2D rectangular plates. The study offers a practical way of understanding the FEM processing in commercial software available in the market and is motivated by the Computer Vibro-acoustic module of the Graduate Program in Mechanical Engineering of the Federal University of Santa Catarina, Brazil. A steel plate with dimensions of 200 mm x 500 mm x 2 mm is implemented, and the results obtained from this algorithm are compared with results given by a commercial software. Two configurations are tested and validated: a free plate, i.e., no boundary conditions, and a door-like plate whose boundary conditions are the door hinges and the door handle. The implementation is made for the direct and superposition methods in FEM, using both symbolic and numerical approaches in MATLAB software. Results obtained present acceptable errors in most frequency bands, thus validating the algorithm and enhancing the understanding of FEM modeling for vibro-acoustic applications.

2aEDb4. K-12 students and Freshmen College Teaching Pattern for Science . Ambika Bhatta (Phys., Lawrence High School, 1 University Ave., Lowell, MA 01854, ambika_bhatta@student.uml.edu), Patrick Nsumeji, and Rafael Cabanas (Phys., Lawrence High School, Lawrence, MA)

This paper presents a model of classroom physics instructions at Health & Human Service (HHS) high school, Lawrence, Massachusetts. Our focus is to create new approaches, procedures, and concepts used in our classrooms to reach the demography. The challenge is to motivate them to do and appreciate science, particularly physics and acoustics. The presented work utilizes the model of audio, video, computer aided and theoretical modalities to help students' access to the fundamental concepts. It will also be shown that it complements any inefficiency in Math and high order thinking skills.

2aEDb5. Teaching ultrasound in air. Craig N. Dolder (Inst. of Sound and Vib. Res., Univ. of Southampton, Highfield Campus, Southampton SO17 1BJ, United Kingdom, C.N.Dolder@soton.ac.uk) and Tim Leighton (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, Hampshire, United Kingdom)

The use of ultrasonics sensors in laboratory exercises and robotics has become popular in recent years. That is, however, only one avenue for exploring the world of inaudible sound that is around us. This talk discusses exercises to interact with the ultrasonic world surrounding us using smartphones, computers, and tablets. Messages can be sent through analog or digital means and the very concept of what is inaudible varies significantly from one person to another. The use of ultrasound is not limited to range finding but also spans communication and art.

2aEDb6. Conceptual understanding and situational interest of middle-school-aged youth who participate in 4-day soundscape science camp.

Maryam Ghadiri Khanaposhtani (Dept. of Forestry and Natural Resources, Purdue Univ., B066 Mann Hall West Lafayette, IN 47906, ghadiry85@gmail.com), ChangChia James Liu (Educational Studies, Purdue Univ., West Lafayette, IN), Bryan C. Pijanowski (Forestry and Natural Resources, Purdue Univ., West Lafayette, IN), and Daniel Shepardson (Departments of Curriculum and Instruction and Earth and Atmospheric Sci., Purdue Univ., West Lafayette, IN)

The purpose of this study was to investigate how participation in an inquiry-based environmental camp contributed to the conceptual understanding and triggered the situational interest of middle school-aged youth to a new field called "Soundscape Ecology." The focus of this study was to understand how participants were affected cognitively and affectively by primary attributes of the immersive soundscape activities. We used descriptive interpretive approaches and several sources of data from drawing activities, pre-post questionnaires, interviews, observations, and participant

artifacts. Our study showed that participants' conceptual understanding as well as their interest were positively affected by variables such as direct interaction with nature, access to authentic technology, collaborative teamwork, and having choice and control. Our results suggest that scientific field work, combined with opportunities to engage youth in scientific education through the use of authentic tools, has the potential to foster an environment in which participants can better comprehend scientific principles.

2aEDb7. Modal Analysis of a Bamboo Composite I-Beam—Results of a collaborative interdisciplinary project. Eoin A. King (Acoust. Program and Lab, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, eoking@hartford.edu), Sigridur Bjarnadottir, and Hernan Castaneda (Civil Eng., Univ. of Hartford, West Hartford, CT)

Bamboo has the potential to be considered a sustainable alternative for conventional construction materials. Traditionally, bamboo culms were used for structural applications (buildings, bridges) in certain parts of the world. The structural behavior of bamboo culms can be unpredictable due to material variations, therefore unsuitable for structural applications in the United States, for example. In recent years, glue laminated bamboo, constructed from bamboo culms that have been crushed and glued together to form boards, has been developed. These boards maintain the excellent mechanical properties of bamboo (high tensile and compressive strength, excellent ductility) while eliminating some material uncertainty. Research into the feasibility of Glue Laminated Bamboo for structural applications, such as I-Beams, is quite novel, and there are many avenues that must be investigated and validated before the standardization of bamboo as a construction material can occur. One key feature that needs to be assessed is how the material properties will influence the dynamic response of an I-Beam during excitation. This paper presents results of a modal analysis of a bamboo composite I-Beam conducted as a collaborative project between undergraduate civil engineering students and acoustical engineering students.

Session 2aIDb**Interdisciplinary: Neuroimaging Techniques I**

Martin S. Lawless, Cochair

Graduate Program in Acoustics, The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Adrian KC Lee, Cochair

University of Washington, Box 357988, University of Washington, Seattle, WA 98195

Sophie Nolden, Cochair

RWTH Aachen University, Jaegerstrasse 17/19, Aachen 52066, Germany

Z. Ellen Peng, Cochair

Waisman Center, University of Wisconsin-Madison, 1500 Highland Avenue, Madison, WI 53711

G. Christopher Stecker, Cochair

*Hearing and Speech Sciences, Vanderbilt University, 1215 21st Ave South, Room 8310, Nashville, TN 37232***Chair's Introduction—9:15*****Invited Papers*****9:20****2aIDb1. Approaches to pushing the limits of human brain imaging.** Bruce Rosen (Radiology, Massachusetts General Hospital, Bldg. 149, 13th St., Rm. 2301, Charlestown, MA 02129, bruce@nmr.mgh.harvard.edu)

By enabling visualization of physiological processes, “Functional imaging,” broadly defined, has dramatically enhanced our ability to explore and better understand human neuroscience and human disease. fMRI has become the keystone of a broad array of functional imaging methods that are revealing the links between brain and behavior in normal and pathological states. Very high strength magnets and advanced large N phased-array coils now enable ultra-high spatial and temporal resolution MRI and fMRI, while advances in MR gradient coil technology have improved our ability to assess tissue microstructure and connectivity almost an order of magnitude. Beyond MRI, positron emission tomography (PET) imaging provides the means to map neurochemical events with exquisite sensitivity, and recent work suggests the potential to extend neurochemical mapping towards quantification of receptor trafficking, and measurements of metabolism and neurotransmitter release dynamics on time frames of a few minutes; tomographic optical imaging allows for portable, bedside assessment of hemodynamics and oxidative metabolism; and densely sampled whole-head magnetoencephalography can, when combined with fMRI, permit high temporal resolution mapping of both cortical and now subcortical brain activity.

10:00**2aIDb2. Finding and understanding cortical maps using neuroimaging.** Martin I. Sereno (Dept. of Psych., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182, msereno@sdsu.edu)

Much of the neocortex, as well as many parts of the brainstem, are divided into “areas” that contain internal topological maps of receptor surfaces. Previously, it was only possible to find the borders and internal organization of these areas using invasive microelectrode mapping and post-mortem architectonics studies in animals. Advances in non-invasive neuroimaging methods over the past two decades have made it possible to extend these studies to the living human brain. This talk summarizes the development and current state-of-the-art of computational methods for non-invasive neuroimaging of cortical maps and cortical areas in human brains, focusing on cortical-surface-based functional MRI, structural MRI, and diffusion-MRI neuroimaging analysis methods originally introduced by my laboratory more than two decades ago. Although topological maps of receptor sheets (e.g., retina, skin touch receptors, and cochlea) are often associated with the earliest stages of sensory processing in the brain, topological maps have turned up in many “higher level” areas. In describing these findings, we emphasize the fundamental architectural bifurcation between visual and somatosensory maps, which are based on a 2D receptor sheet, and auditory maps, which are based on a 1D line of receptors.

10:40–11:00 Break

11:00

2aIDb3. Neuroimaging of the speech network. Frank Guenther (Boston Univ., 677 Beacon St., Boston, MA 02115, guenther@cns.bu.edu)

Historically, the study of the neural underpinnings of speech has suffered from the lack of an animal model whose brain activity could be measured using invasive electrophysiological techniques. The development of non-invasive structural and functional neuroimaging techniques in the latter part of the 20th century has led to a dramatic improvement in our understanding of the network of brain regions responsible for speech production. Techniques for measuring regional cerebral blood flow, including positron emission tomography (PET) and functional magnetic resonance imaging (fMRI), have illuminated the neural regions involved in various aspects of speech, including feedforward control mechanisms as well as auditory and somatosensory feedback control circuits. More recently, fMRI studies utilizing repetition suppression have been used to identify the neural representations used in different parts of the speech network, including the identification of a syllable representation in left ventral premotor cortex. Magnetic resonance imaging has also been used to investigate the anatomical structure of the speech network, providing crucial information regarding connectivity within the network as well as identifying anomalies in the sizes of neural regions and/or white matter pathways in speech disorders.

11:40

2aIDb4. Using electroencephalography as a tool to understand auditory perception: Event-related and time-frequency analyses. Laurel Trainor (Psych., Neuroscience & Behaviour, McMaster Univ., 1280 Main St. West, Hamilton, ON L8S4K1, Canada, ljt@mcmaster.ca)

Electroencephalography (EEG) largely reflects postsynaptic field potentials summed over many (hundreds of thousands of) neurons that are aligned in time and orientation. These electrical fields propagate in all directions such that determination of the sources of electrical fields measured at the surface of the head is much less accurate than localizations using fMRI. However, EEG can be measured with sub-millisecond timing resolution, offering a great advantage for studies of hearing. EEG can measure activity from various nuclei along the subcortical pathway, from primary and secondary auditory cortex and from cortical regions beyond. EEG can be particularly useful for understanding preconscious processing stages, and auditory processing in infants and others who can not make verbal responses. Traditional methods of analysis relate peaks (“components”) in the EEG time waveform to stages of processing. However, communication between brain circuits is reflected in neural oscillations, which can be measured through time-frequency analyses of EEG recordings. Such approaches reveal, for example, how frequency is encoded in the brainstem, and how predictive timing and predictive coding are accomplished in the cortex. I will illustrate these points with example applications largely from our lab and argue that EEG can greatly enhance our interpretation of psychophysical data.

2a MON. AM

MONDAY MORNING, 26 JUNE 2017

ROOM 200, 9:15 A.M. TO 12:20 P.M.

Session 2aMU

Musical Acoustics and Psychological and Physiological Acoustics: Session in Memory of David Wessel

William M. Hartmann, Cochair

Physics and Astronomy, Michigan State University, Physics-Astronomy, 567 Wilson Rd., East Lansing, MI 48824

Andrew C. Morrison, Cochair

Natural Science Department, Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431

Chair's Introduction—9:15

Invited Papers

9:20

2aMU1. David Wessel—The Michigan State years. William M. Hartmann (Phys. and Astronomy, Michigan State Univ., Physics-Astronomy, 567 Wilson Rd., East Lansing, MI 48824, hartmann@pa.msu.edu)

After finishing his Ph.D. work with Estes at Stanford, David Wessel came to Michigan State as an assistant professor of psychology. Dave arrived with a strong background in mathematical psychology and with an intense passion for making music with computers. In the 1970s, Dave and I presented two electronic music concerts to standing-room-only audiences. The division of our labors was straightforward. I provided the venue and audio system. Dave provided all the music, which he was able to do because of his many connections

with electronic musicians around the world. Dave also organized the first computer music conference, held in East Lansing during one of the worst snow storms of the year. This talk will also describe Dave's initial studies of musical timbre and his early years at IRCAM, as I observed them from East Lansing and in Paris.

9:40

2aMU2. David Wessel—The IRCAM years. Tod Machover (Media Lab, MIT, MIT Media Lab, 75 Amherst St., Cambridge, MA 02139, tod@media.mit.edu)

David Wessel was a musical visionary who combined scientific rigor, technological savvy, and sonic adventure to powerfully influence the formative years of Pierre Boulez's IRCAM in Paris. Wessel was trained in mathematics and percussion, receiving a Stanford Ph.D. in musical psychoacoustics. He brought these specialties to computer music at the crucial moment when real-time digital synthesis was being developed. His *Antony* (1977) was the first musical work to use Giuseppe di Giugno's 4A machine, and his *Timbre Maps* (1978) demonstrated for the first time that sonority alone could produce structural relationships. Wessel brought free-jazz principles to live computer music performance, and was a pioneer in understanding and influencing the development of MIDI. Wessel became IRCAM's Director of Pedagogy in 1980, and in that role inspired a generation of international composers, technologists and scientists, a veritable Who's Who of today's most prominent creators. As a member of IRCAM's Artistic Committee, Wessel influenced the selection of artists for IRCAM residences and helped to invent a successful model for combining pedagogy, research and creation. Above all, David Wessel's omnivorous love of all kinds of music, and his deep generosity, brought an unequalled spark of humanity to the world of man, music, and machines.

10:00

2aMU3. David Wessel's Inventive Directorship of UC Berkeley's Center for New Music and Audio Technologies (CNMAT). Adrian Freed (1608 MLK Jr. Way, Berkeley, CA 94709, adrian@adrianfreed.com)

The main professional focus of David Wessel's final 30 years was the development and nurturing of UC Berkeley's Center for New Music and Audio Technologies (CNMAT). Jean-Baptiste Barrière succinctly described David Wessel as bringing a scientific consciousness to music and a musical consciousness to science. This was manifest in CNMAT practice by building apparatus/instruments that served musical production AND research-apparatus that was concurrently validated as novel and significant in three communities: music, science and engineering. I present the major achievements of CNMAT and the special transdisciplinary practices that made the center so productive for its modest size in its 3 concurrent spheres: research, music creation, and education. This will include a brief case study of CNMAT's unique acoustics research apparatus, a 141-driver spherical speaker array. Larger institutions attempted unsuccessfully to create such a high resolution array. David Wessel led CNMAT's success by attracting strong researchers and support engineers over an extended period, creatively finding funding from a diverse combination of extra-mural, government and industry sources, bringing together experts from multiple institutions internationally and tapping the intellectual capital of the UC Berkeley academic community. I conclude by pointing out recent initiatives of David's mentees who carry CNMAT practices in their work.

10:20

2aMU4. David Wessel—A unique professor in Berkeley. Ervin Hafter (Psych., Univ. of California, Berkeley, 1854 San Lorenzo Ave., Berkeley, CA 94707, hafter@berkeley.edu)

Among my best experiences during a half century in Berkeley was the opportunity to interact as friend and colleague of Professor David Wessel. His remarkable blend of scientific brilliance and creativity allowed him to look deeply into questions, figure out the good bits, and come up with new and exciting approaches to an answer. This connection between theory and solution defined his role on a student's committees, and I found that both students and their advisors were grateful for the clarity of his advice. If you asked David for help, you found a seemingly endless fount of generosity, a gift that made him special to everyone who worked with him. Today, I will reminisce on an array of memories that range from his depth of knowledge in both the sciences and arts, on his skills as a chef, on his willingness to provide technical help to everyone, on his compulsion to get us interested in new forms of music, and even on his charming mid-western-American accented French. Yes, Wessel was a brilliant guy, but we will also remember him as a particularly sweet person who left a mark on those fortunate enough to have known him.

10:40–11:00 Break

11:00

2aMU5. David Wessel: A few stories of an antidisciplinarian. Psyche Loui (Wesleyan Univ., 207 High St., Middletown, CT 06459, ploui@wesleyan.edu)

Timbre, gesture, Open Sound Control, additive synthesis, parallel computing: These were just a few of David Wessel's many brainchildren. As a Professor of Music, a founding director of Center for New Music and Audio Technologies (CNMAT), and an Affiliate Professor of Psychology at UC Berkeley, David Wessel was a wise advisor and a wonderful scientist, musician, and friend. I will narrate the exceptional experience of working with the creative force that was David Wessel, both from the perspective of a music perception and cognition scientist, and through the lens of Berkeley's CNMAT, which thrived under his leadership as a synergistic center for performers, researchers, and composers.

Contributed Paper

11:20

2aMU6. Pitching timbre analogies with David Wessel. Punita G. Singh (Sound Sense, 16 Gauri Apartments, 3 Rajesh Pilot Ln., New Delhi 110011, India, punita@gmail.com)

Contemporary thinking and research on timbre and its use as a dynamic, structural component in music performance have been profoundly influenced by the insights and insounds of David Wessel. His intrepid and creative approach opened up vistas of timbre spaces navigable through multidimensional trajectories. Wessel's experiments with timbre streaming [Computer Music J. 3, 4552, (1979)] inspired my own work on perceptual organization of complex-tone sequences [Singh, J. Acoust. Soc. Am. 82,

886-899 (1987)]. The finding of a timbre "interval" akin to a pitch interval as a threshold for streaming reinforced Wessel's notion of timbral analogies [Ehresman and Wessel, IRCAM Rep 13/78, (1978)]. Later work on measuring timbre differences through FO thresholds for streaming (Singh and Bregman, J. Acoust. Soc. Am. 102(4), 1943-1952, (1997)) also lent support to the idea of intervallic relationship between timbres. More recently, my work relating Auditory Scene Analysis to Hindustani rhythms brought us together again, presenting and drumming in a multicultural percussion session at the ASA meeting in San Francisco in 2013. For a person so into timing and timbre, David's untimely departure dealt a discordant blow that can be partially assuaged through such tributes that review, extend, and honor his work.

Invited Papers

11:40

2aMU7. David Wessel—A scholar and a performer. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

In addition to his scholarly work in many fields related to the fields of music psychology and computer music, David Wessel was a gifted composer and a talented percussionist. In this presentation, I will mention my brief interactions with David as we looked at the acoustics of a unique musical instrument: the hand-played hang. The hang is an instrument inspired by the Caribbean steelpan which was popular for a time with many percussionists. After a few comments, we will remember David's musical legacy with some videos of his performances.

12:00–12:20 Panel Discussion

Session 2aNSa

Noise, Architectural Acoustics, and ASA Committee on Standards: Noise Impacts and Soundscapes on Outdoor Gathering Spaces I

Brigitte Schulte-Fortkamp, Cochair

Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 101789, Germany

K. Anthony Hoover, Cochair

McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362

Chair's Introduction—9:15

Invited Papers

9:20

2aNSa1. Tranquillity in the city—Building resilience through identifying, designing, promoting, and linking restorative outdoor environments. Greg Watts (Eng. and Informatics, Univ. of Bradford, Chesham, Richmond Rd., Bradford, West Yorkshire BD7 1DP, United Kingdom, g.r.watts@bradford.ac.uk)

Tranquil spaces can be found and made in the city and their promotion and use by residents and visitors is an important means of building resilience. Studies have shown that spaces that are rated by visitors as tranquil are more likely to produce higher levels of relaxation and less anxiety that should ultimately result in health and well-being benefits. Such spaces can therefore be classed as restorative environments. Tranquil spaces are characterized by a soundscape dominated by natural sounds and low levels of man-made noise. In addition, the presence of vegetation and wild life has been shown to be an important contributory factor. Levels of rated tranquillity can be reliably predicted using a previously developed model TRAPT and then used to design and identify tranquil spaces, improve existing green spaces and develop Tranquillity Trails to encourage usage. Tranquillity Trails are walking routes that have been designed to enable residents and visitors to reflect and recover from stress while receiving the benefits of healthy exercise. By way of example three Tranquillity Trails designed for contrasting areas are described. Predictions of the rated tranquillity have been made along these widely contrasting routes. Feedback from users was elicited and used to gauge benefits.

9:40

2aNSa2. Acoustic renovation for an office courtyard near a busy highway. Bennett M. Brooks and Nathaniel Flanagin (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com)

A complex of several office buildings utilizes a common courtyard as an outdoor gathering space. Regularly scheduled events and celebrations occur in this space. These activities can be disrupted by the noise emitted by vehicular traffic on a nearby busy highway. Field tests were conducted to quantify and characterize the background ambient sound due to the road traffic. The office complex, courtyard, and highway were modeled in a computer aided design system, and various renovation concepts to reduce the perception of highway noise at the venue were studied. The auralized results of design studies were used to generate a virtual reality presentation for evaluation by office management. The reduction of highway noise by several treatment options was noticeable. Also, perceived voice clarity (speech intelligibility) in the courtyard improved with reduced noise. The recommended design treatment provided a significant calculated reduction in highway noise level, with an improved acoustic environment (as perceived) for comfort and speech clarity.

10:00

2aNSa3. Attention focusing in complex acoustical environments. Sabrina Skoda (Inst. of Sound and Vib. Eng., Düsseldorf Univ. of Appl. Sci., Münsterstraße 156, Düsseldorf 40476, Germany, sabrina.skoda@hs-duesseldorf.de), Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Berlin, Germany), André Fiebig (HEAD Acoust. GmbH, Herzogenrath, Germany), and Jörg Becker-Schweitzer (Inst. of Sound and Vib. Eng., Düsseldorf Univ. of Appl. Sci., Düsseldorf, Germany)

The prediction of the perceived overall sound quality of environments consisting of multiple sound sources poses a challenge. The interaction of different sound events results in a great amount of sensory information to be processed but human cognitive capacity is limited. Therefore, listeners tend to focus attention on specific events, the choice of which is influenced not only by the available environmental information but also by the current tasks being performed and individual conditions. To investigate how human attention can be drawn to singular sound sources in complex environments and how this affects the overall evaluation, a series of listening experiments was carried out at Duesseldorf University of Applied Sciences. Participants were asked to evaluate the sound quality of different acoustical environments which consisted of varying combinations of environmental sounds. The most noticeable sound events were identified and were individually rated in the same evaluation. The results show that the overall pleasantness in a complex acoustical environment can be well explained based on the ratings of singular environmental sounds.

10:20

2aNSa4. A methodology to awake citizens' awareness on the effects of leisure noise. Luigi Maffei, Massimiliano Masullo, Giuseppe Ciaburro, and Luigi D'Onofrio (Dept. Architecture and Industrial Design, Università degli Studi della Campania "Luigi Vanvitelli," Via S.Lorenzo, Aversa 81031, Italy, luigi.maffei@unina2.it)

In recent years, several cities' administrations have proposed projects of urban renewal of historical districts to face state of neglect and degradation problems. Frequently these projects include new outdoor social activities such as pubs, bistros, and shops along and/or on the streets with a huge amount of investments, and the success of these projects is then measured through the increase in local population and tourists that frequent, during day and night time, these sites. On the other hand, more people frequent these sites more crowd noise and more consequent complaints of the resident population can be expected. Finally, the due imposition of administrative restrictions can nullify the original aim of the project. Starting from these considerations, a methodology based on sound recordings of different outdoor gathering spaces, on subjective questionnaires administered during listening tests, on features extraction's algorithms, and finally on the artificial neural networks is proposed. The methodology aims to improve the awareness of citizens about the impacts that leisure noise may have on a specific urban project and in general on the quality of the urban environments.

10:40

2aNSa5. In-situ measurements of soundscapes at outdoor gathering spaces—How to consider reliability aspects. André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de)

It is well acknowledged that soundscape investigations must be carried out in the original context. Original contexts like outdoor gathering spaces show usually a strongly varying behavior and are highly uncontrolled. Therefore, if outdoor spaces with highly time-variant noise conditions are considered, it appears difficult to measure reliably the status quo of the site under scrutiny, reflecting measurement uncertainties appropriately. When and how long must be measured, how must residents or visitors be interviewed to obtain reliable and valid data are emerging questions in the context of soundscape investigations. To investigate the reliability of acoustic as well as perceptual measurement data gained by in-situ measurements at specific outdoor gathering spaces, consecutive acoustic and perceptual measurements were performed in Aachen and analyzed. Based on these measurements, some basic requirements to be met regarding reliability and validity were derived and will be discussed. Such investigations are relevant for the preparation of the second part of the International Standard on Soundscapes ISO 12913-2 dealing with data collection.

11:00

2aNSa6. Nestled in nature and near to noise—The Ford Amphitheatre. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The historic Ford Amphitheatre was relocated in 1920, from what would later become the location of the Hollywood Bowl, across the Hollywood Freeway, into an arroyo with a dramatic natural backdrop. Since then, the freeway has become increasingly noisy. The original wood structure was destroyed by a brush fire in 1929, and rebuilt in concrete in 1931, with several subsequent modifications and renovations. The current major renovation to this 1200-seat outdoor amphitheatre includes an expanded "sound wall" that will help to mitigate freeway noise while providing optimal lighting and control positions. The remarkably uniform distribution of ambient noise throughout the seating area and the apparent contributions by the surrounding arroyo will be discussed, along with some of the unique design and construction opportunities.

11:20

2aNSa7. Recovering historic noise. Pamela Jordan (Eng. Acoust., Tech. Univ. Berlin, Technische Universität Berlin, Institut für Strömungsmechanik und Technische Akustik Fachgebiete der Technischen Akustik, Sekr. TA7, Einsteinufer 25, Berlin 10587, Germany, pam.f.jordan@gmail.com)

It is easy to think of noise like refuse—an undesired byproduct of other activities to be minimized or eliminated. However, the richest source of information about a historic site is frequently the garbage that individuals and groups have left behind. Applying the same logic, non-designed sound can also be perceived as an essential component of a heritage location, providing a sensorial understanding of past realities as well as contemporary conditions. Could noise be approached as a resource rather than simply a dilemma, even in cities? This paper seeks to reframe the concept of noise in urban environments with a focus on outdoor heritage sites. The Berlin Wall will be presented as a case study where visitors' perception of unintentional sound provides key information about the past. Inadvertent preservation of historic noise sources and patterns along the Wall, such as vehicular traffic, landscape maintenance, and visitor crowds, has enabled visitors to experience the soundscape of the past in situ rather than through a recording or secondary source. By extending valuations beyond the present moment, it is possible to see the potential value in all sounds.

11:40

2aNSa8. Lessons learning from successful projects in Soundscapes of outdoor spaces. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

One of the most successful projects within Soundscapes is the redevelopment of the Nauener Platz in Berlin. Integrating the Soundscape approach from the beginning of the project enabled a horizontal, long-term dialog with the people in the area. The project that resulted was effectively guided by many participants, resulting in a unique solution for mitigating noise and creating a much-needed "backyard" for the local residents through an improved soundscape. Evaluation followed up confirms the long-term positive effect of the project. The temptation with this level of success is to apply the strategies from Nauener Platz wholesale to other locations and attempt to replicate its achievements. However, this would be a false promise, even at seemingly very similar sites. An instructive example is the Berlin Wall Memorial, which shares many physical attributes to the Nauener Platz. However, its political and historical layers provide a very different set. The paper will discuss the similarities and differences with regard to the ISO.

12:00

2aNSa9. Soundscape analysis and modeling of outdoor gathering spaces. Gary W. Siebein, Hyun Paek, Marylin Roa, Gary Siebein, and Keely Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

Case studies of three outdoor gathering spaces will be presented to illustrate the soundscape concepts embodied in each. Case study 1 is an outdoor amphitheater that is located near a residential neighborhood. Concerts at the amphitheater raised concerns from residents about acoustical measuring methods. Case study 2 is a lively restaurant with a large outdoor seating area where guests eat and listen to music played by a single performer or small group. Case study 3 is a series of outdoor restaurants that do not have live entertainment. The use of different types of soundwalks to capture the acoustical signature of these venues; the acoustical communities involved in each situation; taxonomies of the sounds that occur at each; the specific acoustical events that comprise the ambient sound in each case; acoustical measurements and modeling used in the analysis of each venue; the extent of the acoustical rooms for performing and listening; an acoustical calendar; the design of the acoustical interventions; and live experiments to document ranges of conditions to residents are documented and discussed.

MONDAY MORNING, 26 JUNE 2017

ROOM 202, 9:15 A.M. TO 12:20 P.M.

Session 2aNSb

Noise, Physical Acoustics, ASA Committee on Standards, and Structural Acoustics and Vibration: Sonic Boom Noise II: Mach Cutoff, Turbulence, 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—9:15

Invited Papers

9:20

2aNSb1. The Aerion AS2 and Mach cut-off. Jason Matisheck (Aerion Corp., 5190 Neil Rd., Ste. 500, Reno, NV 89502, jrmatisheck@aerioncorp.com)

The Aerion supersonic business jet is intended to fly at Mach cut-off conditions over land. We will outline the history of investigations into Mach cut-off flight. Aerion's concept of operations for Mach cut-off will be discussed. We will discuss the current state of the technology and identify areas where further research is required.

9:40

2aNSb2. Preliminary assessment and extension of an existing Mach cut-off model. Zhendong Huang and Victor Sparrow (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, zfh5044@psu.edu)

Nicholls' Mach cut-off theory is re-examined for suitability and extensibility. Nicholls' analytical predictions (from 1971) determined the cut-off Mach numbers based on the local speed of sound, along with wind speed and direction at flight elevation and locations where the acoustic rays bend parallel to the ground's surface. The current investigation seeks to understand the limitations of the original formulation and to develop modified predictions to account for realistic and continuous atmospheric profiles. In this presentation, several differences between Nicholls' work and an extended ray tracing model are highlighted. The influence of atmospheric profiles and model assumptions on cut-off Mach numbers is examined. [Work supported by the FAA. The opinions, findings, conclusions, and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

10:00

2aNSb3. Sensitivity analysis of supersonic Mach cut-off flight. Gregory R. Busch, Jimmy Tai, Dimitri Mavris, Ruxandra Duca, and Raathesvar Mohan (Aerosp. Eng., The Georgia Inst. of Technol., 275 Ferst Dr. NW, Atlanta, GA 30332, gbusch3@gatech.edu)

Supersonic aircraft designers are pursuing various methods to help facilitate the re-introduction of overland supersonic flight operations. A substantial amount of research has been invested over recent years to demonstrate its feasibility. An alternative method for satisfying the noise standards for supersonic aircraft is more operation-oriented. Under Mach cut-off conditions, the vehicle still generates sonic booms but the acoustic waves refract in such a way that it does not reach the ground. To better understand the propagation of sonic booms during Mach cut-off flight, Georgia Tech (GT) has conducted research under the FAA's Aviation Sustainability Center (ASCENT). An acoustical model for Mach cut-off flight was developed—GT leveraged this model for sensitivity analysis. The Mach cut-off model allowed GT to vary both atmospheric and flight conditions to study how these dynamic parameters impact sonic boom signatures through the atmosphere. The results of these analyses provide greater insight on how Mach cut-off flight can be achieved and highlights potential technologies to facilitate its re-introduction. [This work was supported by the FAA. The opinions, findings, conclusions, and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

10:20–10:40 Break

10:40

2aNSb4. Subjective study on attributes related to Mach-cutoff sonic booms. Nicholas D. Ortega, Michelle C. Vigeant, and Victor Sparrow (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, njo5068@psu.edu)

Mach cut-off occurs under certain flight conditions (aircraft speed, atmospheric conditions, etc.) in which sonic booms do not reach the ground's surface. The theory has been around for over 40 years, but no studies have investigated the perception of such sounds. The goal of the present investigation is to develop a vocabulary related to the perception of these sounds, with the intent on using this vocabulary to study annoyance. The current study uses sounds recorded by NASA's "Far-field Investigation of No-boom Thresholds" (FaINT) tests. Subjects listened to sets of Mach cut-off audio recordings and were asked to provide descriptors to characterize the sounds. Subjects then listened to each recording and assigned descriptors from their individual set of attributes to each sound, effectively categorizing the sounds. The analysis was performed across subjects to refine definitions and identify commonalities. The vocabulary is used in a follow-up study wherein participants rate their annoyance to a variety of Mach cut-off sounds. The subject test data and subsequent analyses are discussed. [Work supported by the FAA. The opinions, findings, conclusions, and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

11:00

2aNSb5. Laboratory test bed for sonic boom propagation. Michael Bailey, Wayne Kreider, Barbrina Dunmire (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, UW, 1013 NE 40th St., Seattle, WA 98105, mike.bailey.apl@gmail.com), Vera A. Khokhlova, Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., CIMU, APL, Univ. of Washington, Seattle, WA), Julianna C. Simon, and Victor W. Sparrow (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Varying ethanol concentration with depth, Hobaek *et al.* [AIP Proc. 838, 2006] simulated the atmospheric sound speed profile scaled in a water tank and mimicked sonic boom propagation. Some limitations of their groundbreaking design included complexity, fragility, size, safety, one-dimensionality, tonal bursts instead of impulsive shocks, and the inability to scale density, absorption, turbulence, or other nonlinearities. Recognizing the time and cost savings of scaled overflight experiments in a laboratory environment, this work is meant to address certain limitations and investigate the Mach cut-off phenomena. Our primary advances are to use gel layers instead of fluid mixtures and lithotripter shock pulses. We defocused and modified a Dornier Compact S lithotripter to vary source angle in a 500-liter polycarbonate tank, measured shocks with a fiber optic hydrophone, and developed fluid-like gels with negligible shear-wave generation and robustness against cavitation. The sound speed, attenuation, inhomogeneity, and nonlinearity are adjustable within each gel layer. The result is a more flexible, controllable, and durable atmospheric analogue. [Work supported by the FAA. The opinions, findings, conclusions, and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

11:20

2aNSb6. Sonic boom near-field predictions and its impact on the accuracy of ground signature predictions. Hao Shen (Boeing Res. & Technol., The Boeing Co., 325 James S McDonnell Blvd., MC S306-4030, Hazelwood, MO 63042, hao.shen@boeing.com)

The widely adopted sonic boom prediction process contains two steps: the near-field signature prediction from a 3D supersonic flow solver, and the ground signature prediction from a quasi-one-dimensional far-field propagation model following the acoustic ray path. The far-field propagation prediction has to start at a near-field distance sufficiently far away from the airplane flight path to ensure accuracy from the quasi-one-dimensional model. A recent study indicates that the widely adopted practice of using 2-3 body length from the flight path is not sufficient for all airplane configurations. Using a larger near-field distance, however, creates a serious challenge to the CFD based near-field prediction method which will suffer with either degradation in solution accuracy or sharp increase in computational cost, or both. A newly developed space marching approach for high fidelity and high efficiency near-field prediction is revisited here. It is capable of providing high accuracy solution at large near-field distance with a small fraction of computational cost required by conventional CFD solvers. It is applied here to study the sensitivity of the ground signatures to the near-field signature distance for a notional low sonic boom airplane configuration to find the optimal near-field signature distance.

2a MON. AM

11:40

2aNSb7. Numerical simulation of sonic boom focusing and its application to mission performance. Sriram Rallabhandi and James W. Fenbert (AMA Inc., Rm. 190-25, Mailstop 442, NASA Langley Res. Ctr., Hampton, VA 23681, sriram.rallabhandi@nasa.gov)

This paper describes the effort undertaken to include numerical boom focusing on a lossy Burgers equation based sonic boom propagation tool called sBOOM. Traditional ray acoustics may break down during acceleration or other aircraft maneuvers that cause the ray tube area to approach zero. The paper describes a way to numerically predict focused sonic boom signatures using both Gill-Seebass similitude and the solution to the non-linear Tricomi equation that models the physics of boom focusing. The paper would then use this capability to determine focusing and non-focusing climb trajectories and their impact on mission performance.

12:00

2aNSb8. International Civil Aviation Organization Supersonic Task Group overview and status. Sandy R. Liu and Bao Tong (Noise Div., Federal Aviation Administration, 800 Independence Ave., SW, Washington, DC 20591, sandy.liu@faa.gov)

A resurgence of interest in supersonic flight has emerged due to recent technological advances. Given the global impact of such aircraft, environmental standards and recommended practices (SARP) are being developed under the International Civil Aviation Organization (ICAO) to ensure the mitigation of accompanying sonic booms prior to any re-introduction of civil supersonic flight operations. In 2004, ICAO's Committee of Environmental Protection (CAEP) established the supersonics work program under Working Group 1 (WG1) to develop SARPs. Current proposed concepts include (1) traditional sonic boom aircraft (like the Concorde), which operate supersonically over water; (2) low-boom designs with overland operation capabilities; and (3) traditional aircraft flying under Mach cut-off conditions. To-date, WG1/SSTG continues to formulate an enroute sonic boom SARP for civil supersonic airplanes. Metrics, test procedures, and a data framework continue to be investigated for suitability and efficiency. A newly formed Landing and Take-Off (LTO) noise subgroup for supersonic aircraft has been tasked to define a second SARP for the terminal environment (i.e., subsonic operations). An overview of ICAO's hierarchy supporting SARPs development, WG1/SSTG technical program, and a current status update is presented.

MONDAY MORNING, 26 JUNE 2017

ROOM 210, 9:20 A.M. TO 12:20 P.M.

Session 2aPAa

Physical Acoustics: Infrasound I

Roger M. Waxler, Cochair

NCPA, University of Mississippi, 1 Coliseum Dr., University, MS 38677

Pieter Smets, Cochair

R&D Department of Seismology and Acoustics, KNMI, PO Box 201, De Bilt 3730 AE, Netherlands

Invited Papers

9:20

2aPAa1. Seismo-acoustic wavefield of strombolian explosions at Yasur volcano, Vanuatu, using a broadband seismo-acoustic network, infrasound arrays, and infrasonic sensors on tethered balloons. Robin S. Matoza (Dept. of Earth Sci., Univ. of California, Santa Barbara, Webb Hall MC 9630, Santa Barbara, CA 93106, matoza@geol.ucsb.edu), Arthur Jolly (GNS Sci., Avalon, New Zealand), David Fee (Univ. of Alaska, Fairbanks, Fairbanks, AK), Richard Johnson (GNS Sci., Avalon, New Zealand), Bernard Chouet, Phillip Dawson (U.S. Geological Survey, Menlo Park, CA), Geoff Kilgour, Bruce Christenson (GNS Sci., Avalon, New Zealand), Eslie Garaebiti (Vanuatu Meteorol. and Geohazards Dept., Port Vila, Vanuatu), Alex Iezzi (Univ. of Alaska, Fairbanks, Fairbanks, AK), Allison Austin (Dept. of Earth Sci., Univ. of California, Santa Barbara, Santa Barbara, CA), Ben Kennedy, Rebecca Fitzgerald, and Nick Key (Univ. of Canterbury, Christchurch, New Zealand)

Seismo-acoustic wavefields at volcanoes contain rich information on shallow magma transport and subaerial eruption processes and inform our understanding of how volcanoes work. Acoustic wavefields from eruptions are predicted to be directional, but sampling this wavefield directivity is challenging because infrasound sensors are usually deployed on the ground surface. We attempt to overcome this observational limitation using a novel deployment of infrasound sensors on tethered balloons in tandem with a suite of dense ground-based seismo-acoustic, geochemical, and eruption imaging instrumentation. We conducted a collaborative multiparametric field experiment at the active Yasur volcano, Tanna Island, Vanuatu, from 26 July to 2 August 2016. Our observations include data from a temporary network of 11 broadband seismometers, 6 single infrasonic microphones, 7 small-aperture 3-element infrasound arrays, 2 infrasound sensor packages on tethered balloons, an FTIR, a FLIR, 2 scanning Flyspecs, and various visual imaging data; scoria and ash samples were collected for petrological analyses. This unprecedented dataset should provide a unique window into processes operating in the shallow magma plumbing system and their relation to subaerial eruption dynamics.

9:40

2aPAa2. Acoustical localization, reconstruction, signal, and statistical analysis of storm electrical discharges from a 2-months long database in southern France. François Coulouvrat (Institut Jean Le Rond d'Alembert (UMR 7190), Université Pierre et Marie Curie & CNRS, Université Pierre et Marie Curie, 4 Pl. Jussieu, Paris 75005, France, francois.coulouvrat@upmc.fr), Thomas Farges, Louis Gallin (CEA, DAM, DIF, Arpajon, France), Arthur Lacroix (CEA, DAM, DIF, Paris, France), and Régis Marchiano (Institut Jean Le Rond d'Alembert (UMR 7190), Université Pierre et Marie Curie & CNRS, Paris, France)

Infrasound and low frequency sounds are discussed as a method to characterize lightning flashes in a complementary way to electromagnetic (EM) observations. Thunder and EM data result mainly from a 2-months long observation campaign in Southern France, dedicated to monitor atmospheric electricity as part of the hydrological cycle in Mediterranean (HyMeX program). Possibilities and limitations to follow storms by sound or infrasound (in the 1 to 40 Hz frequency range) at various distances are outlined. The influence of distance, wind, and ambient noise is examined. Several examples of individual lightning flashes acoustical reconstruction are compared to EM reconstruction by means of a Lightning Mapping Array. Both Intra-Cloud or Cloud-to-Ground (CG) discharges are investigated. Special emphasis is brought to the lower part of CGs, as many acoustic signals are localized inside the lightning CG channel. A statistical comparison between the acoustical versus EM approaches is performed, thanks to a significant number of recorded discharges in a single storm. Performances of acoustical reconstruction are detailed as function of observation range. Detailed signal analysis compared to a theoretical model shows that the tortuous channel geometry explains at least partly the low frequency content of our observations of thunder spectra.

10:00

2aPAa3. Infrasound and internal gravity waves generated by atmospheric storms. Sergey Kulichkov, Igor Chunchuzov, Oleg Popov, Vitaly Perepelkin, and Elena Golikova (Obukhov Inst. of Atmospheric Phys., 3 Pyzhevsky Per., Moscow 119017, Russian Federation, snk@ifaran.ru)

The recordings of infrasound and internal gravity waves (IGWs) obtained during 2015-2016 at infrasound station I43 IMS and a network of microbarographs installed by Obukhov Institute of Atmospheric Physics (OIAP) are presented. The OIAP network of microbarographs is capable of detecting simultaneously an infrasound at frequencies less than 3 Hz and IGWs with periods ranging from 5 min to 3 hr. It is shown that the low-frequency wave processes generated by atmospheric fronts retain high coherence (0.6-0.9) over the areas with horizontal dimensions of a few tens of km. It is found that after the passage of atmospheric front the internal wave trains were observed with the amplitudes considerably exceeding those of IGWs that were detected before the passage of the atmospheric front. The discrete periods of 35 min, 56 min, and 110 min were found in the frequency spectra of the observed wave trains. For these periods, the coherence between atmospheric pressure variations measured at different points reach local maxima, and the sum of the phase differences between selected three points tends to zero. The phase speeds for the observed IGWs are in the range of 10-50 m/c. The wave "precursors" with amplitudes of 10 Pa and periods of 15-20 min were also detected 10 to 15 hr before a passage of the atmospheric front through a network. Along with IGWs, the infrasound associated with atmospheric front was also detected (August 2016) by I43 in the frequency range 0.1-0.4 Hz.

10:20–10:40 Break

10:40

2aPAa4. Infrasound and low frequency sound emitted from tornados. Carrick L. Talmadge (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38655, clt@olemiss.edu)

The NCPA, in collaboration Hyperion Technology Group, has performed a series of measurements of infrasound and low-frequency sound generated by tornadic thunderstorms near Oklahoma City, OK, during a large-scale outbreak on May 24, 2011. Ground truth for tornado tracks as well as meteorological data were available for these storms. Infrasound and low-frequency sound were identified separately for a long duration EF-5, an EF-4, and an EF-2 tornado. As reported by Frazier et al. [JASA **135**, 1742 (2014)], infrasound in two distinct regions were noted: An infrasound band between approximately 1-10 Hz and a low-frequency audible band located between roughly 40-200 Hz (center frequency around 80-100 Hz). As part of the NOAA VORTEX-SE initiative, the NCPA will be collecting additional infrasound data in the Northern portion of Alabama, centered on Huntsville and the Sand Mountain region. Current plans are to install to 10 infrasound arrays, with seven elements per array. As part of the same VORTEX-SE initiative, seven additional arrays will be deployed by the University of Alabama Huntsville. We will report here on the status of infrasound generated by tornadic thunderstorms, as well as discuss the status of modeling efforts to understand the origins of these emissions.

11:00

2aPAa5. Acoustic characterization of a portable infrasound source. Martin Barlett, Thomas G. Muir, Charles M. Slack, and Timothy M. Hawkins (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713, barlett@arlab.utexas.edu)

A trailer-able, pneumatic infrasound source is described that can produce tones with frequencies as low as 0.25 Hz. The device is based on compressed gas air flow which is released into the atmosphere from a pair of 500 gallon reservoirs, pressurized to 200 psi (1377 kPa) and modulated by pair of rotating ball valves, with 2 in. (5 cm) diameter ports. Positive air flow is released twice per revolution, so the device is a siren. In addition to the fundamental frequency, the siren also produces a series of harmonics of the fundamental tone, enabling simultaneous measurements to be made at multiple frequencies, up to 20 Hz. This instrument was developed to support in-situ calibrations of infrasound sensors and arrays as well as to provide a controllable source for other infrasound studies, such as signal insertion loss measurements of wind noise suppression structures and characterization of infrasound array azimuthal directivity. The physical attributes of the source are described and results of acoustic measurements of sound pressure levels and azimuthal directivity are presented. The results are also compared to estimates made using improved versions of a previously presented aero-acoustic model (Pneumatic Infrasound Source: Theory and Experiment, POMA 19, 045030 [2013] and papers 4pPA3 and 4pPA4. J. Acoust. Soc. Am. 134 [2013]). [Work supported by the U.S. Army Space and Missile Defense Command/Army Forces Strategic Command (USASMD/ARSTRAT).]

2a MON. AM

11:20

2aPAa6. Reciprocity calibration for infrasound sensors. Thomas B. Gabrielson (Penn State Univ., PO Box 30, State College, PA 16804, tbg3@psu.edu)

Frequency response is a key parameter in understanding the impact of a transducer on a measurement. A single volts-per-pascal value is often cited as an acoustic transducer's response; however, the magnitude and phase of this ratio over the entire frequency range of interest is required to understand the effects of that transducer on input waveforms. While this ratio can be determined by comparison to a reference transducer, any reference must itself be calibrated in some fashion. The National Center for Physical Acoustics has built two calibration chambers designed for evaluation of infrasound microbarometers. The large interior volume (about 1.5 cubic meters) allows simultaneous testing of several microbarometers and reference transducers. These chambers are also equipped with two drivers—10-in. subwoofers—so that two-tone linearity testing can be done. The incorporation of two drivers opens the possibility for implementing reciprocity calibration, a well-established primary calibration methodology. This paper describes development, evaluation, and uncertainties of a reciprocity-based calibration procedure designed expressly for measuring the complex frequency response of infrasound sensors in the 0.005 to 10 Hz frequency range.

11:40

2aPAa7. Wind noise reduction using a compact infrasound sensor array and a Kalman filter based on the Matérn Covariance Function. William G. Frazier (Hyperion Technol. Group, Inc., 3248 West Jackson St., Tupelo, MS 38804, gfrazier@hyperiontg.com)

A method for real-time estimation of stationary infrasound signals such as microbaroms in wind noise at low signal-to-noise ratios using a compact infrasound sensor array is presented. A compact array is defined as a sensor array that has an aperture that is much smaller than the shortest infrasound wavelengths of interest and is unsuitable for estimation of direction-of-arrival. In this application, the spacing between sensors results in the measured wind noise being highly correlated, and therefore, simple averaging cannot be used to obtain a good estimate of the infrasound signal. However, by adequately modeling the spatiotemporal wind noise process, array gain can be realized. The method is based on using a Kalman Filter that is designed with the assumption that the measured wind noise can be adequately modeled as a dynamic Gaussian random field with a Matérn covariance function (demonstrated previously at the ASA Meeting in Salt Lake City, Utah, May, 2016). The presentation describes how to design the Kalman Filter in order to estimate the infrasound signal of interest, demonstrates the filter performance using synthetic and measured data from a compact array, and describes how to extend the method to support changing wind conditions and non-stationary infrasound signals.

12:00

2aPAa8. Association of impulsive infrasonic events at medium ranges. W. C. Kirkpatrick Alberts and Stephen M. Tenney (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20723, william.c.alberts4.civ@mail.mil)

Multiple, widely spaced, infrasonic arrays are routinely used to detect and localize impulsive events of unknown origin at medium ranges (<100 km). Event data are subsequently processed to yield line of bearing (LOB) information and localization is accomplished manually. This method of analysis could significantly benefit from automatic association and localization. Because infrasound arrays are often separated by many tens of kilometers and signals reaching the arrays can be significantly altered along the propagation path, the task of associating signals is difficult and time consuming. Further, confidence in an event association is difficult to assign to a signal due to arrival timing and local interferers. By using beamforming methods and coherence between signals, it is possible to automatically associate a given recorded event. At each array, a delay and sum beamformer is used to calculate the LOB to an unknown source. The delayed and summed beam at each array is then used to calculate the pairwise coherence between all beams. Impulsive events due to sources recorded by widely spaced infrasound arrays often exhibit high coherence at many of the frequencies in the signal. Examples of successful associations between widely spaced arrays will be discussed.

Session 2aPAb

Physical Acoustics: General Topics in Physical Acoustics I

Alexey S. Titovich, Chair

Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd., West Bethesda, MD 20817

Contributed Papers

10:20

2aPAb1. Active diffracting gratings for vortex beam generation in air. Ruben D. Muelas H., Jhon F. Pazos-Ospina, and Joao L. Ealo (School of Mech. Eng., Universidad del Valle, Ciudad Universitaria Meléndez. Bldg. 351., Cali 760032, Colombia, joao.ealo@correounivalle.edu.co)

Vortex beams VB have gained great attention because of their interesting features, e.g., autorreconstruction ability, capacity to transport and transfer angular momentum, among others. Different applications have been proposed, e.g., particle manipulation and rotation control of particles/objects. Recently, VB generated in water using passive structures have been reported. In particular, multi-arm spiral slits are used attached to a radiating source. In this work, we propose a new alternative to generate VB in air. Specifically, we use active diffracting gratings easily fabricated by gluing a ferroelectret film on a lower electrode, structured on a PCB, that perfectly resembles the desired geometry. The active material is not cut to size and shape. See [1]. Consequently, a transducer with the geometry of the intended grating radiates the acoustic energy. Broadband spiral active gratings are employed to create VB in air at frequencies between 100 and 200 kHz. Numerical simulations are compared with experimental results. This new class of transducers paves the way to the creation of complex radiation fields in a rapid, cheap, and efficient manner. [1] J. Ealo, J. Camacho, and C. Fritsch, "Airborne ultrasonic phased arrays using ferroelectrets: a new fabrication approach," *IEEE Trans. Ultrason. Ferroelectr. Freq. Control* 56(4), 848-858 (2009).

10:40

2aPAb2. Experimental study of the angular momentum transfer capacity of grating vortex beams. Ruben D. Muelas H., Joao L. Ealo, and Jhon F. Pazos-Ospina (School of Mech. Eng., Universidad del Valle, Bldg. 351, Cali, Colombia, ruben.muelas@correounivalle.edu.co)

Acoustic vortices (AV) are special beams with a screw-type dislocation, a pressure null along their principal axis and the ability to carry angular momentum. Mono-element sources, metamaterials, and phased arrays can be used to generate an AV. Among them, phased arrays are employed in the most of works that report particle manipulation with AV because of their ease to shape beams. However, special care must be taken at the design stage of the array because additional zones of constructive interference may appear, i.e., the so called grating lobes surrounding the main beam. When generating vortex beams using phased arrays, we have experimentally verified that these grating lobes have similar characteristics than the main beam, so we called them grating vortices (GV). However, GV exhibit certain level of distortion in both phase and magnitude. In this work, we present an experimental study of the angular momentum transfer capability of GV of different topological charge. We estimate the mechanical torque induced and the angular momentum transferred to disk-like samples of different size. The vortices are generated using a 30-element array operating at 40 kHz in air. A discussion on the potential of use of GV in the manipulation of particle/objects is given.

11:00

2aPAb3. A three-dimensional simulation of vortex formation at the open end of an acoustic waveguide. Carlos Malaga (School of Sci., Universidad Nacional Autonoma de Mexico, Ciudad Universitaria, AV. UNIVERSIDAD N° 3000, Facultad de Ciencias, Mexico City, Mexico City 04510, Mexico, cmi.ciencias@ciencias.unam.mx), Leon Martinez (CCADET, Universidad Nacional Autonoma de Mexico, Mexico City, Mexico), Roberto Zenit (IIM, Universidad Nacional Autonoma de Mexico, Mexico City, Mexico), and Pablo L. Rendon (CCADET, Universidad Nacional Autonoma de Mexico, Mexico City, Mexico)

For high enough levels of acoustic pressure inside a tube, a nonlinear mechanism is responsible for the formation of annular vortices at the open end of the tube, which results in energy loss. Higher sound pressure levels in the tube lead in turn to larger values of the acoustic velocity at the exit, and thus to higher Reynolds numbers. It has been observed [Buick et al., 2011] that two regimes are possible depending on whether the acoustic velocity is low, in which case vorticity appears in the immediate vicinity of the tube, or high, in which case vortices are formed at the open end of the tube and are advected outwards. We use a Lattice Boltzmann Method (LBM) to simulate the velocity field at the exit of the tube in 3D, for both cases. We plan to compare these numerical results with experimental results obtained through particle image velocimetry (PIV). The effect of varying both the geometry of the tube and the shape of the termination on the magnitude of the nonlinear losses at the exit is also examined.

11:20

2aPAb4. Three-dimensional model for acoustic field created by a piezoelectric plate in a resonator. Goutam Ghoshal, Benjamin P. Ross-Johnsrud, Kedar C. Chitale (FloDesign Sonics, 380 Main St., Wilbraham, MA 01095, g.ghoshal@fdsonics.com), Yurii A. Ilinskii, Evgenia A. Zabolotskaya (Appl. Res. Labs, Univ. of Texas at Austin, Austin, TX), and Bart Lipkens (FloDesign Sonics, Springfield, MA)

A three-dimensional model is developed to describe an acoustic field excited by a piezoelectric plate of finite size in a fluid filled resonator. First, the eigenfunctions (modes) of a bare plate are derived using general piezoelectric equations considering the elastic and electric properties of the plate. Then, the piezoelectric plate is placed into a fluid media such that only one plate side is in fluid and an acoustic field generated by the plate in the fluid is estimated. Finally, a reflector is placed to be parallel to the piezoelectric plate and acoustic field in a resonator is evaluated. The solution for a piezoelectric plate of finite size is obtained using Singular Value Decomposition (SVD) method. Equations for acoustic and electric variables are presented. Radiation force on spherical particles in the standing wave field is derived and discussed. Numerical results are presented to show the three-dimensional modal displacement and electrical characteristics of the plate at various frequencies and aspect ratio. Finally, the analytical results are compared with two-dimensional and three-dimensional finite element results using COMSOL Multiphysics commercial software.

11:40

2aPAb5. Directed self-assembly of three-dimensional user-specified patterns of particles using ultrasound. Milo Prisbrey, John Greenhall (Mech. Eng., Univ. of Utah, 201 Presidents Circle, Salt Lake City, UT 84119, mprisim@gmail.com), Fernando Guevara Vasquez (Mathematics, Univ. of Utah, Salt Lake City, UT), and Bart Raeymaekers (Mech. Eng., Univ. of Utah, Salt Lake City, UT)

Particles dispersed in a fluid medium are organized into three-dimensional (3D) user-specified patterns using ultrasound directed self-assembly. The technique employs standing ultrasound wave fields created by ultrasound transducers that line the boundary of a fluid reservoir. The acoustic radiation force associated with the standing ultrasound wave field drives the particles into organized patterns, assuming that the particles are much smaller than the wavelength, and do not interact with each other. A direct solution method is theoretically derived to compute the ultrasound transducer operating parameters required to assemble a user-specified pattern of particles in any 3D simple, closed reservoir geometry with any arrangement of ultrasound transducers. This method relates the ultrasound wave field and its associated radiation force to the ultrasound transducer operating parameters by solving a constrained optimization problem that reduces to eigendecomposition. Experimental validation of the method is accomplished by assembling 3D patterns of carbon nanoparticles in a cubic water reservoir lined with four ultrasound transducers. Additionally, the versatility of the method is demonstrated by simulating ultrasound directed self-assembly of complex 3D patterns of particles in cubic and noncubic reservoir geometries lined with many ultrasound transducers. This method enables employing

ultrasound directed self-assembly in a variety of engineering applications, including biomedical and materials fabrication processes.

12:00

2aPAb6. Waves in non-conducting continuum with frozen-in magnetization. Victor Sokolov (Dept. of Mathematics, Moscow Technol. Univ., Av. Vernadskogo 78, Moscow 119454, Russian Federation, vvs195326@gmail.com)

The report reviews the new approach to the ferrohydrodynamics and magnetoelasticity, based on the concept of frozen-in magnetization. Up till now, there were well-known two frozen-in vector fields. The first one is the vorticity field was introduced by Helmholtz. The second frozen-in field is a magnetic field in a perfectly conducting fluid was introduced by Alfven. The acoustic approximation of the ferrohydrodynamic equations with frozen-in magnetization allowed us to describe the experimental results on the anisotropy of the ultrasonic velocity in magnetized magnetic nanofluids and predict new waves: Alfven-type hydrodynamic wave and slow magneto-sonic one. Alfven-type wave is accompanied by oscillations of the magnetization. It is shown that, in a non-conducting solid with a frozen-in magnetization, the propagation of three types of waves are possible: the longitudinal wave, pure shear wave, and new wave which have mixed form of Alfven-type and shear waves. The theoretical results are found to agree well with the experimental data on the dependence of the velocity of longitudinal and transverse waves in polycrystalline nickel on the magnetizing field strength. The set of dynamical equations that we have derived can be used to tackle many problems in ferrohydrodynamics and magnetoelasticity.

MONDAY MORNING, 26 JUNE 2017

ROOM 311, 9:15 A.M. TO 11:20 A.M.

Session 2aPPa

Psychological and Physiological Acoustics and Speech Communication: Acoustics Outreach to Budding Scientists: Planting Seeds for Future Clinical and Physiological Collaborations

Anna Diedesch, Cochair

Dept. of Otolaryngology/Head & Neck Surgery, Oregon Health & Science University, 3710 SW US Veterans Hospital Rd., Portland, OR 97239

Adrian KC Lee, Cochair

University of Washington, Box 357988, Seattle, WA 98195

Chair's Introduction—9:15

Invited Papers

9:20

2aPPa1. Microsecond interaural time differences of acoustic transients are decoded by inhibitory-excitatory interactions in neurons of the lateral superior olive. Tom P. Franken (KU Leuven; The Salk Inst. for Biological Studies, SNL-R, 10010 N Torrey Pines Rd., La Jolla, CA 92037, tfranken@salk.edu), Philip H. Smith (Univ. of Wisconsin-Madison, Madison, WI), and Philip X. Joris (KU Leuven, Leuven, Belgium)

The lateral superior olive (LSO) in the auditory brainstem generates sensitivity to interaural level differences (ILD), an important cue for sound localization, by comparing excitatory (E) input from the ipsilateral ear with inhibitory (I) input from the contralateral ear. Large axosomatic synapses (e.g., calyx of Held) point to the importance of precise temporal processing, but that is not easily reconciled with ILD detection. We propose that the IE interaction allows detection of interaural time differences (ITD) of acoustic transients, to

which humans are exquisitely sensitive. We obtained *in vivo* whole cell recordings of LSO and MSO neurons in the gerbil, while presenting monaural and binaural clicks. We found that ITD functions to clicks in the LSO are surprisingly steep, in contrast to MSO neurons, which are considered the main ITD detectors. Intracellular LSO recordings show EPSPs generated by the ipsilateral click and IPSPs by the contralateral click, where IPSPs often arrive earlier. Binaural spiking is maximally suppressed when the EPSP coincides with the falling slope rather than the peak of the IPSP. We conclude that LSO neurons are more sensitive to ITDs of transients than MSO neurons. This clarifies the importance of timing specializations in the LSO circuit.

9:40

2aPPa2. Quantifying connectivity to auditory cortex: Implications for crossmodal plasticity and hearing restoration. Blake E. Butler and Stephen G. Lomber (Psych., Univ. of Western ON, Social Sci. Bldg., 11, London, ON N6A5C2, Canada, bbutler9@uwo.ca)

When one sensory modality is lost, compensatory advantages are observed in the remaining senses. There is evidence to suggest these advantages reflect recruitment of cortical areas that normally process sound. In the cat, crossmodal reorganization of auditory cortex appears to be field specific. While little or no activity is evoked in primary cortical regions by visual and somatosensory stimulation, higher-level fields confer increased peripheral acuity and improved visual motion detection. In order to better understand the changes in neural connections that underscore these functional adaptations, we have undertaken a series of detailed anatomical studies aimed at quantifying and comparing the patterns of connectivity in hearing and deaf animals. A retrograde neuronal tracer was deposited into auditory cortical areas, coronal sections were taken, and neurons showing positive retrograde labeling were counted and assigned to cortical and thalamic areas. Projections within and between sensory modalities were quantified; while some small-scale differences emerge, patterns of connectivity are overwhelmingly preserved across experimental groups within each cortical field examined. This structural preservation has implications for our understanding of the mechanisms that underlie crossmodal reorganization; moreover, it suggests that the connectivity necessary for resumption of auditory function may withstand even lengthy periods of deprivation.

10:00

2aPPa3. Dynamic emergence of categorical perception of voice-onset time in human speech cortex. Neal P. Fox (Dept. of Neurological Surgery, Univ. of California, San Francisco, Sandler Neurosci. Bldg., UCSF Mission Bay, 675 Nelson Rising Ln., Rm. 510, San Francisco, CA 94143, neal.fox@ucsf.edu), Matthias J. Sjerps (Dept. of Linguist, Univ. of California, Berkeley, Nijmegen, Netherlands), and Edward F. Chang (Departments of Neurological Surgery and Physiol., Univ. of California, San Francisco, San Francisco, CA)

A fundamental challenge in speech perception involves the resolution of a many-to-one mapping from a highly variable, continuous sensory signal onto discrete, perceptually stable categories that bear functional relevance. Recent work has identified signatures of invariance in early neural responses to speech, but the physiological mechanisms that give rise to these categorical representations remain unclear. We employed intracranial recordings in human subjects listening to and categorizing speech stimuli to investigate the spatio-temporal cortical dynamics underlying categorical perception. Stimuli comprised a voice-onset time (VOT) continuum from /b/ (0 ms VOT) to /p/ (50 ms VOT). Results revealed spatially distinct neural populations that respond selectively to tokens from one category (either /b/ or /p/). Within these subpopulations, response amplitude is modulated by stimulus prototypicality for within-category stimuli (e.g., stronger response to 0 ms vs. 10 ms VOT in /b/-selective electrodes). Over the course of a trial, this initially graded encoding of VOT rapidly evolves to reflect properties of the ultimate (categorical) behavioral response function. These same dynamics emerged in a computational neural network model simulating neuronal populations as leaky integrators tuned to detect temporally distributed acoustic cues at precise lags. Our results provide direct evidence that categorical perception of VOT arises dynamically within discrete, phonetically tuned neural populations.

10:20

2aPPa4. Disrupted auditory nerve activity limits peripheral but not central temporal acuity. Carol Q. Pham and Fan-Gang Zeng (Ctr. for Hearing Res., Univ. of California Irvine, 110 Med Sci E, Irvine, CA 92697, carol.pham@uci.edu)

Auditory neuropathy affects synaptic encoding or neural conduction of signals in the cochlea or the auditory nerve. Subjects with auditory neuropathy poorly recognize speech in noise which correlates with poor temporal processing. The integrity of temporal processes in the auditory system can be assessed with detection of just-noticeable differences in gap between sounds. Disorder in the auditory periphery appears to alter the precise timing or latency of synchronous neural discharges important for temporal coding. However, the relative contribution of auditory nerve activities to central temporal processing is unknown. Auditory neuropathy produced significantly worse than normal gap detection within a frequency but normal gap detection between different frequencies. No correlation between same- and different-frequency gap detection supports two temporal processes: a peripheral mechanism dependent on overlapping nerve fibers mediating same-frequency gaps and a central mechanism dependent on cross-correlated activity of non-overlapping fibers mediating different-frequency gaps. The fast, peripheral mechanism enables temporal acuity on the order of milliseconds and is likely limited by neural synchrony, the amount of total nerve activity, or both, whereas the sluggish, central mechanism is likely limited by switching time between perceptual channels on the order of a hundred milliseconds. The results demonstrate auditory nerve activities limit peripheral but not central temporal acuity.

10:40

2aPPa5. Electro-oculography based horizontal gaze tracking: A perspective for attention-driven hearing aids. Lubos Hladek, W. Owen Brimijoin (MRC / CSO Inst. of Hearing Res. (Scottish Section), Glasgow Royal Infirmary, 10-16 Alexandra Parade, New Lister Bldg. 3L, Glasgow Royal Infirmary, Glasgow G31 2ER, United Kingdom, lubos.hladek@nottingham.ac.uk), and Bernd Porr (School of Eng., Univ. of Glasgow, Glasgow, United Kingdom)

Users of hearing aids with standard directional microphones can have difficulties in complex listening situations such as multi-talker environments because they must turn their heads to follow conversations that move rapidly from one talker to the next. The use of gaze-pointed hearing aid directionality has been suggested as a way to potentially alleviate this problem [Kidd *et al.* (2013); J. Acoust. Soc.

Am. 2013; 133(3): EL202-7.] However, to arrive at a practical and usable device, there is the need for unobstructive and mobile technology for gaze tracking. Here, we propose and evaluate an algorithm for estimating eye-gaze angle based solely on the single channel electro-oculogram (EOG), which can be obtained directly from the ear canal using conductive hearing aid molds. In contrast to conventional techniques, we use an algorithm which calculates the absolute eye angle by statistical analysis of the saccades. This results in robust long term performance where predicted eye angles significantly correlate with actual eye angles. This opens up the possibility of an attention driven beam-former for hearing aids without the need for eye-tracking goggles. [This work was supported by the Medical Research Council [grant number U135097131], Chief Scientist Office (Scotland) and Oticon Foundation.]

11:00

2aPPa6. Selecting an acoustic correlate for automated measurement of /r/ production in children. Heather M. Campbell (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., Fl. 6, New York, NY 10012, heather.campbell@nyu.edu), Daphna Harel (Ctr. for the Promotion of Res. Involving Innovative Statistical Methodology, New York Univ., New York, NY), and Tara McAllister Byun (Communicative Sci. and Disord., New York Univ., New York, NY)

A current need in the field of speech pathology is the development of reliable and efficient techniques for the purpose of evaluating changes in speech production over the course of treatment. The industry standard for scoring speech is time consuming and expensive, as it involves aggregating perceptual ratings across expert listeners. As techniques for automated measurement of speech improve, acoustic measures have the potential to play an expanded role in the clinical management of speech disorders. The current study asks which of several acoustic measures of children's productions of English /r/ corresponds most closely with ratings given by trained listeners. This study fits a series of ordinal mixed effects regression models to a large sample of children's /r/ productions that had previously been rated by three trained listeners (speech-language pathologists). Controlling for age, sex, and allophonic contextual differences, the acoustic measure that accounted for the most variance in speech rating was F3-F2 distance, normalized relative to a sample of age- and gender-matched speakers. Therefore, this acoustic measure is recommended for use in future automated scoring of children's production of rhotic targets.

MONDAY MORNING, 26 JUNE 2017

ROOM 311, 11:40 A.M. TO 12:20 P.M.

Session 2aPPb

Psychological and Physiological Acoustics: Models and Reproducible Research I

Alan Kan, Cochair

University of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705

Piotr Majdak, Cochair

Acoustics Research Institute, Austrian Academy of Sciences, Wohllebengasse 12-14, Wien 1040, Austria

Invited Papers

11:40

2aPPb1. The challenges in developing useful models. Barbara Shinn-Cunningham and Le Wang (Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu)

Models can be extremely helpful in understanding the mechanisms governing hearing, providing insights into how information is processed and combined to enable perception and communication. Yet modeling is complicated and is more of an art than a science. Ideally, a model should include only those key components that are critical for describing known phenomena. Yet, models are typically under-constrained, so that modelers constantly are forced to make educated guesses based on limited data. Given this, developing a useful model requires good intuition and an eye for what is essential and what is superfluous: a useful model requires navigating a balance between realism/complexity versus tractability/interpretability. Verification is often only indirect, by using the resulting model to generate testable predictions, and comparing these predictions to new experimental data. Moreover, when a model fails (such as when predictions do not match empirical outcomes), the path to "fixing" the model is not always straightforward or clear. Some of these challenges will be illustrated by our own recent efforts to model envelope following responses in the brainstem.

12:00

2aPb2. On the use of hypothesis-driven reduced models in auditory neuroscience. Dan F. Goodman (Elec. Eng., Imperial College London, London SW7 2AZ, United Kingdom, d.goodman@imperial.ac.uk)

There are a number of detailed models of auditory neurons that are able to reproduce a wide range of phenomena. However, using these models to test hypotheses can be challenging, as they have many parameters and complex interacting subsystems. This makes it difficult to investigate the function of a mechanism by varying just one parameter in isolation, or to assess the robustness of a model by systematically varying many parameters. In some cases, by limiting the scope of a model to testing a specific hypothesis using a particular set of stimuli, it is possible to create a reduced mathematical model with relatively few, independent parameters. This has considerable advantages with respect to the problems above. In particular, if a certain behavior is robust and does not depend on finely tuned parameters, then different implementations are more likely to produce the same results—a key property for reproducible research. In addition, the code for these models is typically simpler and therefore more readable, and can often run faster, enabling us to carry out systematic parameter exploration. I will illustrate these points with a reduced model of chopper cells in the ventral cochlear nucleus.

MONDAY MORNING, 26 JUNE 2017

ROOM 201, 9:15 A.M. TO 12:20 P.M.

Session 2aSAa

Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics: Acoustic Metamaterials I

Christina J. Naify, Chair

Acoustics, Naval Research Lab, 4555 Overlook Ave. SW, Washington, DC 20375

Chair's Introduction—9:15

Invited Papers

9:20

2aSAa1. Acoustic metasurfaces with rapid change profiles. Chengzhi Shi, Marc Dubois, Yuan Wang, and Xiang Zhang (Dept. of Mech. Eng., Univ. of California, Berkeley, 3112 Etcheverry Hall, Berkeley, CA 94720, chengzhi.shi@berkeley.edu)

Acoustic metasurfaces with subwavelength unit cells modulating the phase of sound wave were developed to realize beam forming, steering, focusing, and recently carpet cloaking. However, these metasurfaces were used for small phase gradient applications and the ability to cloak or mimic reflection surfaces with rapid change profiles remained unexplored. Here, we analytically and experimentally investigate the effect of rapid change in-plane phase gradient on the design of acoustic metasurfaces for carpet cloaking and diffusing applications. Helmholtz resonators with different neck radii are used to form the ultrathin metasurface in our study. The variation of neck radius of the Helmholtz resonator results in different reflection phase of acoustic wave. In both analytical and experimental results, the large in-plane phase gradient does not affect the carpet cloak design, which is solely depending on the phase correction of the local profile height. On the contrary, the impedance mismatch resulted from the rapid change profile yields additional scattering to the reflection from such rapid change profiles. The recreation of the reflection pattern of rapid change profiles from a flat surface using metasurface needs to consider this multiple scattering, which requires additional amplitude modulation from the metasurface.

9:40

2aSAa2. Metamaterial-based manipulation of orbital angular momentum for sound. Bin Liang, Xue Jiang (Dept. of Phys., Inst. of Acoust., Nanjing Univ., P. R. China, 22 Hankou Rd., Nanjing, Jiangsu 210093, China, liangbin@nju.edu.cn), Jiajun Zhao (Dept. of Phys. and Ctr. for Nonlinear Dynam., Univ. of Texas at Austin, Austin, TX), Jianchun Cheng (Dept. of Phys., Inst. of Acoust., Nanjing Univ., P. R. China, Nanjing, China), Likun Zhang (Dept. of Phys. and Ctr. for Nonlinear Dynam., Univ. of Texas at Austin, Austin, TX), and Chengwei Qiu (Dept. of Elec. and Comput. Eng., National Univ. of Singapore, Singapore, Singapore)

The existing wave-steering devices generally control acoustic beams carrying only linear momentum. But acoustic vortices imprinted with orbital angular momentum (OAM) obviously opens new degree of freedom to control wavefront, with wide applications such as in the design of acoustic “screwdrivers” capable to generate a rotation torque to rotate objects contactlessly. To overcome the drawbacks of the traditional methods for producing acoustic OAM, we have designed and fabricated an acoustic vortex emitter with multi-arm coiling slits that utilize diffraction effect to generate vortex beam, with broadband functionality and stable topological charge. Based on this, we further propose a new mechanism for producing acoustic OAM by converting acoustic resonances to OAM. As an implementation, we have designed and fabricated a thin planar device with high efficiency to verify our scheme. Compared with existing

ways for OAM production by phased spiral sources that need sophisticated electronic control and by physically spiral sources that need screw profiles and may also have a bulky size, our acoustic resonance-based OAM production bears the advantages of high efficiency, planar profile, compact size, no spiral structure, and can be freely tuned to produce different orders of OAM. I will also discuss some potential applications of our proposed scheme for OAM manipulation by metamaterials.

Contributed Papers

10:00

2aSAa3. Demonstration of a broadband aqueous acoustic metasurface.

Matthew D. Guild, Charles Rohde, Theodore P. Martin, and Gregory Orris (US Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, mdguild@utexas.edu)

Acoustic metamaterials have been utilized in recent years to demonstrate extreme acoustic properties, such as those with negative or near-zero dynamic values. While effective, the use of acoustic metamaterials can lead to voluminous structures that may not be practical for some applications. Alternatively, ultrathin structures known as acoustic metasurfaces offer the same capability to achieve extreme properties as acoustic metamaterials while offering the added benefit of having negligibly small (i.e., subwavelength) thickness. In this work, we will discuss an aqueous acoustic metasurface that utilizes subwavelength structures designed to acoustically act in parallel, allowing for a thin, modular structure to be realized while achieving a broad range of effective surface properties. A theoretical formulation for the design of the flexural elements is presented, accounting for the elastic motion of the elements subject to fluid loading due to the water. Based on this design, an aqueous acoustic metasurface was constructed from a brass plate, which was machined to achieve the prescribed flexural elements on the surface, and experimentally tested in water. The results of this analysis and testing will be discussed. [Work supported by the Office of Naval Research.]

10:20

2aSAa4. Anomalous refraction and asymmetric transmission of SV-waves through elastic metasurfaces. Xiaoshi Su and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, xiaoshi.su@rutgers.edu)

Recent advances in acoustic metasurface design make it possible to manipulate sound waves in an almost arbitrary way. Here, we present several elastic metasurfaces comprised of an array of subwavelength plates for controlling SV-wave in solids. The underlying physics are the coupling between the SV-wave in the elastic body and the flexural wave in plates, and the coupling between the P-wave in the elastic body and the longitudinal wave in plates. By varying the thicknesses of the plates, a wide range of phase delay for flexural waves can be obtained, while keeping constant the phase delay for longitudinal waves. The anomalous refraction of SV-waves is achieved by selecting the thickness of each plate to engineer the phase change according to the generalized Snell's law. This metasurface has another feature that it redirects SV-waves only, which enables it to be used to split SV- and P-wavefronts into different directions. In addition, this metasurface can be paired with a uniform metasurface to break spatial symmetry and achieve asymmetric transmission for normally incident SV-waves. Other potential applications, such as focusing and negative refraction, will also be discussed. [Work supported through ONR MURI.]

10:40

2aSAa5. Omnidirectional sound shielding with acoustic metacages.

Chen Shen, Yangbo Xie, Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC), and Yun Jing (Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus box 7910, Raleigh, NC 27695, yjing2@ncsu.edu)

Omnidirectional sound barriers are useful for various applications in noise reduction. Conventional sound insulating structures like micro-perfo-

rated plates or porous materials prevent the exchange of airflow. Here, we propose the design of an acoustic metacage which can shield acoustic waves from all directions and have the ability of allowing air pass through freely. The mechanism is that the strong parallel momentum along the surface rejects sound regardless of the directions of the incident wave. Structures based on open channels and Helmholtz resonators are designed at an operation frequency of 2.49 kHz with thickness less than half of the wavelength. A prototype is fabricated using 3D printing and further verified experimentally in a waveguide. Simulation and measurement results clearly show that the proposed metacage can shield acoustic waves when the sources are placed either interior or exterior. An average energy decay of more than 10 dB is achieved when a loudspeaker is placed inside the metacage within a certain frequency band. Our metacage can have applications where ventilation is required.

11:00

2aSAa6. Design of broadband acoustic metamaterials for low-frequency noise insulation. Zibo Liu, Leping Feng, and Romain Rumlper (Dept. of Aeronautical and Vehicle Eng., KTH Royal Inst. of Technol., Stockholm SE-100 44, Sweden, zibo@kth.se)

An innovative configuration of an acoustic sandwich structure is proposed in this paper, which uses the locally resonant structures to generate stopbands in desired frequency regions and hence to increase the sound transmission loss of the panel. Effects of different types of resonators, including the mounting techniques, are investigated. The methods to broaden the effective stopbands are discussed. The acoustic properties of the sandwich panel with non-flat laminates are also studied. Numerical analyses show that good results can be obtained when combining the laminate modification with the locally resonant structure, especially when the stopbands are designed to compensate the corresponding coincidence effects of the sandwich panel. The analysis is based on the Finite Element models constructed in COMSOL. Bloch wave vectors are derived at first Brillouin zone by using wave expansion method. Dispersion relation of the structure is discussed. Experimental validation is planned, and the results will be shown in the conference.

11:20

2aSAa7. Designing beampatterns with tapered leaky wave antennas.

Christina J. Naify (Jet Propulsion Lab., 4555 Overlook Ave. SW, Washington, DC 20375, christina.naify@gmail.com), Katherine. Woolfe (Code 7160, National Res. Council Postdoctoral Associateship, Naval Res. Lab, Washington, DC), Christopher N. Layman (ATS Consulting, Pasadena, CA), Jeffrey S. Rogers, Matthew D. Guild, and Gregory Orris (Code 7160, Naval Res. Lab, Washington, DC)

Leaky wave antennas (LWAs) have been shown to be an effective tool for frequency-steerable wave radiation in both the electromagnetic and acoustic wave regimes. LWA's operate by modifying the impedance on a waveguide such that refraction occurs out of the waveguide at an angle corresponding to Snell's Law. For a LWA with uniform leaking parameter across the waveguide length, that leakage angle is constant. Using analytical techniques, and by careful geometric design of the waveguide impedance, the leaked beampattern can be tailored. The process of the tapering process for an acoustic LWA is discussed here, and notional examples are presented including sidelobe reduction. [Work supported by the Office of Naval Research.]

11:40

2aSAa8. Low frequency bandgaps in lightweight metamaterial panels using rotation inertia multiplication. Tommaso Delpero (Mech. Integrity of Energy Systems, Empa, Dübendorf, Switzerland), Gwenael Hannema, Stefan Schoenwald, Armin Zemp (Acoustics/Noise Control, Empa, Dübendorf, Switzerland), Andrea Bergamini (Mech. Integrity of Energy Systems, Empa, Dübendorf, Switzerland), and Bart Van Damme (Acoustics/Noise Control, Empa, Ueberlandstrasse 129, Dübendorf 8600, Switzerland, bart.vandamme@empa.ch)

Of all possible features of structural metamaterials, the formation of bandgaps is the most studied one due to its direct application for sound and

vibration isolation. While achieving low frequency values for the position of the first bandgap is, in general terms, not an unsurmountable challenge, the combination of material properties such as high stiffness, low density, and reduced size of the unit cell, with low (in absolute terms) frequency bandgaps, may well require some careful consideration. In previous work, we designed panels with a 3D network of resonators, clearly improving the vibration isolation compared to a homogeneous panel with the same weight. Recently, we have devised a novel implementation of inertia amplification, based on coupling the energy of longitudinal waves into the rotational oscillation of inertia elements within the unit cell. In this contribution, we present examples of phononic crystals based on this approach, and we discuss the interaction of acoustic waves with the discussed lattices.

Invited Paper

12:00

2aSAa9. Optimal sound-absorbing structures. Ping Sheng (Dept. of Phys., HK Univ. of Sci. & Technol., Clear Water Bay, Kowloon, Hong Kong 000, China, sheng@ust.hk)

Causal nature of the acoustic response dictates an inequality that relates the absorption spectrum of the sample to its thickness. We use the causal constraint to delineate what is ultimately possible for sound absorbing structures, and denote those which can attain near-equality for the causal constraint to be “optimal.” By using acoustic metamaterial as backing to conventional porous absorbers, a design strategy is presented for realizing structures with target-set absorption spectra and a sample thickness close to the minimum value as dictated by causality. By using this approach, we have realized a 12 cm-thick structure that exhibits broadband, near-perfect flat absorption spectrum starting at around 400 Hz, while the minimum sample thickness as calculated from the causal constraint is 11.5 cm. To illustrate the versatility of the approach, two additional optimal structures with different target absorption spectra are presented. This “absorption by design” strategy enables the tailoring of customized solutions to difficult room acoustic and noise remediation problems. [Work done in collaboration with Min Yang, Shuyu Chen, and Caixing Fu.]

MONDAY MORNING, 26 JUNE 2017

ROOM 204, 9:20 A.M. TO 12:20 P.M.

Session 2aSAb

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

Benjamin Shafer, Chair

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

Contributed Papers

9:20

2aSAb1. Modeling and estimation of focused ultrasound radiation force for modal excitation. Songmao Chen, Christopher Niezrecki, Alessandro Sabato, and Peter Avitabile (Mech. Eng., Univ. of Massachusetts Lowell, One University Ave., Lowell, MA 01854, song.m.chen@gmail.com)

To date, conventional excitation techniques such as impact hammer or mechanical shaker excitations are commonly used in experimental modal testing. However, these excitations require that the excitation device be in direct contact with test articles, resulting in measurement distortions, particularly for small structures like MEMS cantilever. In addition, it is physically difficult or even impossible to apply these contact type excitations to certain structures, for example, biological tissues and thumb nail sized turbine

blade. Moreover, these conventional excitations have limited bandwidth, usually below than 10 kHz, and thus not applicable for structures with interest in higher frequency modes. Focused ultrasound radiation force, having a much broader frequency bandwidth, has recently been used to excite structures with sizes ranging from micro to macro-scale. Therefore, it can potentially be used as an alternative non-contact excitation method for experimental modal analysis. Yet, this force remains to be quantified in order to obtain the force-response relationship, i.e., the frequency response functions (FRFs) of test articles. The dynamic focused ultrasound radiation force is modeled and estimated using the calibrated sound pressure fields generated by a spherically focused ultrasonic transducer (UT) driven by amplitude modulated signals. Its application for modal excitation is to be discussed.

2aSab2. Investigation of various damping measurement techniques.

Christian A. Geweth (Chair of VibroAcoust. of Vehicles and Machines, Tech. Univ. of Munich, Boltzmannstraße 15, Garching b. München 85748, Germany, christian.geweth@tum.de), Patrick Langer (Chair of VibroAcoust. of Vehicles and Machines, Tech. Univ. of Munich, Munich, Bavaria, Germany), Kheirollah Sepahvand (Chair of VibroAcoust. of Vehicles and Machines, Tech. Univ. of Munich, Garching bei Munich, Germany), and Steffen Marburg (Chair of VibroAcoust. of Vehicles and Machines, Tech. Univ. of Munich, Muenchen, Germany)

In order to compare experimentally determined damping values with damping values from simulations, high effort is necessary. A precise modeling of the boundary conditions with respect to its impact on the damping is often difficult to realize and often time consuming. Furthermore, the used measurement parameters, like sampling frequency or windowing can have an unneglectable influence on the experimentally determined damping value. In order to observe the sensitivity of single parameters during the determination of damping, the dynamical behavior of virtual models with a known excitation is investigated. The used numerical methods are validated with analytical solutions for the model. The obtained time data from the model are used to determine the damping with different methods. Therefore, the influence of the different methods on the model can be identified. The virtual modeling opens up the opportunity to identify and quantify different sources of errors and disturbance.

10:00

2aSab3. Spatial distribution of acoustic radiation force modal excitation from focused ultrasonic transducers in air.

Thomas M. Huber, Ian McKeag, William Riihiluoma (Phys., Gustavus Adolphus College, 800 W College Ave., Saint Peter, MN 56082, huber@gac.edu), Christopher Niezrecki, Songmao Chen, and Peter Avitabile (Mech. Eng., Univ. of Massachusetts Lowell, Lowell, MA)

Recent studies have utilized the acoustic radiation force for non-contact modal excitation of structures in air. When two ultrasonic frequencies, for example, $f_1 = 610$ kHz and $f_2 = 600$ kHz, are incident on an object, the acoustic radiation force produces a driving force at a difference frequency $f_1 - f_2 = 10$ kHz. The current study compared the spatial distribution of driving force from a pair of co-focused transducers emitting f_1 and f_2 , to a single focused transducer emitting an amplitude modulated signal of both f_1 and f_2 . The difference frequency ranged from 400 Hz to 80 kHz. Ultrasonic transducers, with focal spot diameters of ~ 2 mm mounted on translation stages, could be directed at a 100kHz PCB-378C01 microphone or a $19.8 \times 6.8 \times 0.37$ mm clamped-free brass cantilever monitored by a Polytec PSV-400 vibrometer. When mixing of frequencies f_1 and f_2 was solely due to the acoustic radiation force, the driving force was localized to a region a few mm in diameter. However, in other cases, very broad spatial distributions of difference frequency excitation were measured; this indicated non-acoustic radiation force mixing of f_1 and f_2 , such as within the transducer. The practical implications for non-contact modal excitation using acoustic radiation force will be discussed.

10:20

2aSab4. Vibro-acoustic modeling of roof panels for analysis of sound radiation from droplet impact.

Sangmok Park, Yunsang Kwak, Deukha Kim, Junhong Park (Hanyang Univ., 222, Wangsimni-ro, Seongdong-gu, Eng. Center, 306, Seoul 04763, South Korea, tkdahr619@hanyang.ac.kr), and Kyungsup Chun (Hyundai Motors, Hwaseong, South Korea)

Sound generated by impacts between raindrops and roof panels of vehicles is an important factor on automotive qualities when driving in rainy conditions. Therefore, analytical method to control this phenomenon is necessary. In this research, a theoretical model for predicting characteristics of sound radiation by droplet impacts was proposed. An experiment for measuring forces generated by falling droplets was conducted. The characteristics of the measured forces were investigated in the frequency domain. A measurement on a plate was performed to understand sound radiation formed by droplet impacts. Correlations between acoustic characteristics and properties of the plate were identified. A vibro-acoustic model was developed to analyze the experimental results. Assuming generation of

sound sources at each location due to the vibrating plate, radiation of sound fields was theoretically calculated and verified by comparing with the measured results. Under single and multi-layered conditions, influence factors on acoustic properties were investigated based on the model. As a result, by using the proposed model, it is possible to predict acoustic mechanisms of vehicles due to raindrops and make them suitable for specific designs.

10:40

2aSab5. Clamping force diagnosis during bolting process using acoustic signatures.

Gyungmin Toh, Jaehong Lee (Mech. Eng., Hanyang Univ., 222, Wangsimni-ro, Seongdong-gu, Seoul 04763, South Korea, avludals@gmail.com), Jaesoo Gwon (Hyundai Motors, Seoul, South Korea), and Junhong Park (Mech. Eng., Hanyang Univ., Seoul, South Korea)

The method of measuring the fastening force of the bolt in a non-contact manner during the fastening of the bolt is of great value in the industry. In this study, the fastening force was measured by changing the dynamic characteristics as the bolts were fastened. Experiments were carried out by measuring the vibration generated when bolts with load cells were fastened. When the bolt is fastened, the vibration characteristics of the bolt are measured by the accelerometer attached to the joint structure of the bolt by the fastener. The measured vibration signals are classified using cepstrum of bolt. Learning was performed by recognizing different axial force as each speaker. The clamping force was predicted in such a manner as to determine which studied clamping force is most similar to the present clamping force. The proposed method is verified by applying it to actual bolt structure.

11:00

2aSab6. A vibroacoustic analysis of pre-stressed saw blades to identify instabilities considering gyroscopic effects and centrifugal forces utilizing the finite element method.

Marcus Guettler (Faculty of Mech. Eng., Tech. Univ. of Munich, Boltzmannstr. 15, Munich 85748, Germany, marcus.guettler@tum.de), Christopher Jelich (Faculty of Mech. Eng., Tech. Univ. of Munich, Garching b. München, Germany), Steffen Marburg (Faculty of Mech. Eng., Tech. Univ. of Munich, Muenchen, Germany), Ettore Grasso, and Sergio De Rosa (Dipartimento di Ingegneria industriale, Università degli Studi di Napoli Federico II, Napoli, Italy)

In various engineering fields, large saw blades on heavy machines are used for several tasks. Typically, in civil engineering, they cut large concrete structures such as walls for adding or changing doors and windows in buildings, as well as they are utilized for roads and bridges maintenance works. In the cutting process, especially large blades tend to vibrate excessively. The unstable behavior can lead to wide cutting lines, less productivity, or even jamming of the saw blade resulting in an unsafe environment for the workers on-site. For these productivity and security related issues, engineers face the challenge to investigate the dynamic behavior of large saw blades at early stages of product development. The finite element method has emerged to be a useful tool when investigating saw blade designs since various effects such as pre-stress and gyroscopic and/or centrifugal forces can be considered. In this work, the authors use the finite element method to study the effect of gyroscopic and centrifugal forces onto pre-stressed saw blades to (i) identify unstable dynamic behavior and further (ii) optimize the design to increase the vibroacoustic stability. In addition, the kinetic energy values are used as a measure for potential sound radiation.

11:20

2aSab7. An experimental investigation into the insertion loss from subscale acoustic enclosures with geometric imperfections.

Christopher Beale, Murat Inalpolat, Christopher Niezrecki, and David J. Willis (Mech. Eng., Univ. of Massachusetts Lowell, One University Ave., Lowell, MA 01854, Christopher_Beale@student.uml.edu)

Enclosures with different geometries constitute the internal sections of various engineering applications including cabins of passenger cars, fuselages of aircraft wings, and internal compartments of wind turbine blades. Acoustic insertion loss from and to these enclosures affect certain objective and subjective acoustic measures along with the ability to detect damage. This presentation describes a thoroughly executed test plan that identifies the effect of geometric imperfections, such as holes, edge splits, and cracks with different severity levels and locations, on the insertion loss from a

subscale acoustic enclosure. A composite rectangular prism enclosure, located inside an anechoic chamber, was internally ensonified using a loudspeaker, and an externally located condenser microphone was used to measure the insertion loss under different conditions. One of the faces of the enclosure possessed various size and location imperfections simulating damage. Insertion loss deviations introduced through the prescribed damage cases were compared to a baseline case with no prescribed imperfections. The results obtained from the initial test campaign with healthy and damaged enclosure specimens were utilized to arrive at several conclusions on the detectability and feature extraction capabilities required for damage detection from subscale enclosures.

11:40

2aSAb8. Simulation of coupled structural-acoustic response with dynamic damage evolution. Jonathan Pitt (Appl. Res. Lab., The Penn State Univ., PO Box 30, Mailstop 3320B, State College, PA 16804, jonathan.pitt@psu.edu)

A novel time-domain method for simulating dynamic damage evolution in a coupled structural-acoustic system is presented. The system is derived via the theory of continuum damage mechanics, and incorporates standard damage evolution models, but is readily extendible to more exotic formulations. The overall solution method is staggered, solving for the dynamic damage evolution first with an explicit step, and then using the new values in the coupled computation of the structural-acoustic system. The spatial domain is discretized using a mixed finite element method, and the temporal domain is discretized with a higher-order implicit time discretization scheme. Efforts toward fully coupled verification of the solution algorithm are presented, as are validation studies for cases without evolving damage. Applications with evolving damage are presented, and present a novel first principles study of changes in the structural acoustic response to

dynamically evolving damage in the structure. Special attention is given to brittle fracture. Examples of downstream usage of the evolving structural response are discussed in the concluding remarks.

12:00

2aSAb9. Using reciprocity principles and sensitivity functions for the vibroacoustic response of panels under random excitations. Christophe Marchetto, Maxit Laurent (Univ Lyon, INSA-Lyon, Laboratoire Vibrations Acoustique, 25 bis av. Jean Capelle, Villeurbanne F-69621, France, christophe.marchetto@usherbrooke.ca), Olivier Robin, and Alain Berry (Groupe d'Acoustique de l'Université de Sherbrooke, Université de Sherbrooke, Sherbrooke, QC, Canada)

The vibroacoustic characterization of panels submitted to random pressure fields is of great interest for the industry. The test means associated with those excitations (i.e., wind tunnel, reverberant room) are expensive and can hardly be controlled. An alternative method to experimentally characterize the behavior of a panel under random pressure fields is therefore proposed. The mathematical formulation of the problem allows describing the vibroacoustic behavior of a panel as a function of the cross spectral density function of the considered excitation and so-called "sensitivity functions." These functions can be estimated experimentally using reciprocity principle, which can either be applied for characterizing the structural response by exciting the panel with a normal force at the point of interest or for characterizing the acoustic response (radiated pressure, acoustic intensity) by exciting the panel with a monopole and a dipole source. For validation purposes, the method is applied numerically and experimentally for the case of a diffuse acoustic field. Based on indicators such as the vibratory response and the transmission loss factor, the method is finally confronted to measurements in coupled anechoic-reverberant rooms facility following standards.

2a MON. AM

MONDAY MORNING, 26 JUNE 2017

BALLROOM A, 9:20 A.M. TO 12:20 P.M.

Session 2aSC

Speech Communication: Speech Production (Poster Session)

Melissa M. Baese-Berk, Chair

Department of Linguistics, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403

All posters will be on display from 9:20 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 9:20 a.m. to 10:50 a.m. and authors of even-numbered papers will be at their posters from 10:50 a.m. to 12:20 p.m.

Contributed Papers

2aSC1. Rhoticity in Cajun French. Katherine M. Blake and Kelly Berkson (Linguist, Indiana Univ., 107 S Indiana Ave., Bloomington, IN 47405, kamblake@indiana.edu)

Previous studies of Cajun French (CF) report a shift in pre-rhotic vowel quality (Conwell & Juilland 1963, Blainey 2015, Dubois & Noetzel 2005, Salmon 2007, Lyche, Meisenburg & Gess 2012). This is unsurprising, in that such behavior is reported for numerous Francophone varieties, but prior work on rhotics in CF has been largely impressionistic. As such, the current study presents acoustic analysis of pre-rhotic vowels in CF. Formant measurements were taken from data collected from three native speakers. The

question of whether rhotic deletion is present in CF is also addressed: /r/-dropping is widely attested cross-linguistically, and has been reported in the closely related Francophone variety of Acadian French (Cichocki 2012, 2006). Given the previous literature on /r/-dropping and rhoticity in CF and other Francophone varieties, it was expected that these data would show variable /r/ production, compensatory lengthening of vowels preceding a dropped /r/, and a general trend of /r/-induced vowel lowering. Results of this study confirm the presence of highly variable rhotic deletion, a lack of compensatory lengthening triggered by this deletion, a centralization effect of the rhotic on the quality of preceding vowels, and variable lowering of the third formant.

2aSC2. Articulatory data for a five-way liquid contrast: 3D ultrasound of Marathi. Kelly Berkson and Abigail H. Elston (Linguistics, Indiana Univ., 1020 E. Kirkwood Ave., Ballantine Hall 844, Bloomington, IN 47405, kberkson@indiana.edu)

Lateral and rhotic consonants show great crosslinguistic variation, and are traditionally described as articulatorily complex (Ladefoged & Maddieson 1996; Proctor 2011; Wiese 2001, 2011). A good body of work has investigated the characteristics of liquids in languages like English (Dellatre & Freeman 1968; Guenther *et al.* 1998; Sproat & Fujimura 1993; Westbury, Hashi, & Lindstrom 1998; many others), which contains a two-way contrast. What of South Asian languages, however, which often contain a greater number of liquids? Tamil liquids have been imaged using palatography and electropalatography (McDonough & Johnson 1997) as well as MRI (Narayanan *et al.* 1999), and Malayalam liquids have been imaged using mid-sagittal ultrasound (Scobbie, Punnoose, & Khattab 2013). Little has been done with Marathi, though. Like Tamil and Malayalam, Marathi—an Indic language spoken in the Indian state of Maharashtra—contains a five-way liquid contrast. This work utilizes recent advances in 3D ultrasonography to provide detailed articulatory data for Marathi's five liquids ($/l/$, $/l^h/$, $/r/$, $/r^h/$, $/l/$) (Dhongde & Wali 2009; Pandharipande 1997). Real-time images of tongue motion are combined with digitized impressions of the palate to provide new insights into the complex articulatory gestures involved in production of these sounds.

2aSC3. Asymmetrical patterns of formant variability in English vowels. Wei-rong Chen, Mark Tiede, and D. H. Whalen (Haskins Labs., 300 George St., Ste. 900, New Haven, CT 06511, chenw@haskins.yale.edu)

Previous studies have claimed that lower formants should be weighted more than higher formants in a perceptual model of vowel perception (e.g., Schwartz *et al.*, 1997). Given this formant weighting hypothesis, and if vowels have acoustic targets, vowels should be more variable in higher formant frequencies. Here, we examined within-speaker variability for five English vowels $/æ, \Lambda, \upsilon, \varepsilon, \iota/$ in various contexts as produced by 32 speakers in the x-ray microbeam database (Westbury, 1994). For variabilities of the first three formants, only $/\upsilon/$ follows this prediction (i.e., variability: $F3 > F2 > F1$) while $/æ/$ exhibits the opposite pattern; if we ignore $F3$ (as being less reliably measured), most vowels conform to the prediction (i.e., variability: $F2 > F1$), except for $/æ/$. Although the $F2$ variability is generally consistent with the perceptual model of vowel perception, it is also consistent with a possibly greater effect of consonant coarticulation (which is extensive here) on $F2$ relative to $F1$; this requires more testing. Further, while these results do not fully conform to the prediction made by the hypotheses, coproduction effects arising from the diverse contexts likely interact with the expected tendency. Correlation with observed kinematic variabilities will also be discussed. [Work supported by NIH grant DC-002717.]

2aSC4. Acoustic properties of Mexico City Spanish vowel weakening. Meghan F. Dabkowski (Dept. of Spanish and Portuguese, The Ohio State Univ., 298 Hagerty Hall, 1775 College Rd., Columbus, OH 43210, dabkowski.5@osu.edu)

Mexico City Spanish exhibits weakened vowels that have been described as reduced, relaxed, unstable, obscured, abbreviated, devoiced, and “lost,” indicating likely reduction in duration, voicing, and/or quality. The objective of this study is to precisely identify the acoustic nature of this vowel weakening. To this end, recorded spontaneous speech was collected from 20 speakers native to Mexico City. 3000 tokens, i.e., monophthongs, were analyzed acoustically in Praat (Boersma & Weenink 2016), and measurements were taken for $F1$, $F2$, vowel duration, and voicing duration. Findings show that vowel weakening in this variety consists primarily of shortening and weakened voicing, but not raising or centralization. Instead of simple presence or absence of voicing, many tokens show weak voicing, characterized by a lower intensity in the waveform and a lighter voice bar, or partial voicing that does not endure throughout the entire vowel. The presence of frication distinguishes devoicing from weak voicing when other aspects of the acoustic signal are not clear indicators. Many tokens exhibited full voicing, but only consisted of 2-3 wave cycles, resulting in a severely shortened vowel. Uncovering the acoustic properties of these weakened vowels is crucial to understanding how this variety fits with cross-linguistic vowel weakening trends.

2aSC5. Vowel acoustics in three dialects of Spanish: Iberian, Dominican, and Mexican. Stephanie C. Fermin, Martha Tyrone (LIU–Brooklyn and Haskins Labs, 1 university Plaza, Brooklyn, NY 10021, Stephanie.c.fermin@gmail.com), Laura L. Koenig (Adelphi Univ. and Haskins Labs., New Haven, CT), and Isabelle Barriere (LIU–Brooklyn, Brooklyn, NY)

A growing number of studies have begun to investigate vowel variability among Spanish speakers. The purpose of this study was to measure the acoustics of vowels in three dialects of Spanish and to compare how these dialects vary in their vowel production. This information is important for speech and language clinicians working with dialectally diverse individuals to recognize the difference between typical dialectal variation and a speech/language disorder. We specifically examined dialects that have developed separately from each other and also have large numbers of speakers. Data were obtained from five female speakers in each of these groups: Iberian speakers, Dominican speakers, and Mexican speakers ($N = 15$). To analyze the effect of speaking task on production variation, we elicited a controlled speaking task and a naturalistic speaking task. For the controlled task, the participants were asked to read aloud randomized phrases on a computer screen. For the naturalistic task, participants were presented with a simple map from which to give navigation instructions. The results showed differences in average vowel placement and token-to-token variability. These data disprove previous hypotheses that vowels are stable across Spanish dialects.

2aSC6. Phonetic variability in Moroccan Arabic rhotics. Aaron Freeman (Dept. of Linguist, Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104, aaronfr@sas.upenn.edu)

Moroccan Arabic $/r/$ and its pharyngealized counterpart exhibit a wide range of variability in their pronunciation, with reported articulations ranging from apical trills to uvular fricatives. Using an phonetic dataset elicited from speakers of the dialect of Fès (reported to traditionally have a uvular variant), I assess the distribution and acoustic properties of rhotic variants, including coarticulatory effects. The data present three distinct rhotic articulations: (1) an apical trill or tap, (2) an apical continuant produced together with high-frequency sibilant noise, and (3) a dorsal sonorant or rhotacized vowel similar to English “burred $/r/$.” Despite claims in the literature, no uvular fricative was observed in the data, and the dorsal sonorant was identified by speakers as the idiosyncratic local pronunciation. Variants (1) and (2) further exhibit devoiced positional variants. Lowered $F2$ of adjacent vowels and of sonorant portions of the rhotic signal differentiate the pharyngealized phoneme $/r^h/$ from its plain counterpart $/r/$. However, both phonemes were found to exhibit the same range of variation in their primary articulation. Two acoustic properties common to all variants were (a) depression and attenuation of upper formants and (b) the presence of some aperiodic noise above 5 kHz.

2aSC7. On the acoustic cues of unreleased stops. Ting Huang (Dept. of Linguist and Philosophy, Massachusetts Inst. of Technol., Graduate Inst. of Linguist, Rm. B306, HSS Bldg., No. 101, Section 2, Kuang-Fu Rd., Hsinchu City 30013, Taiwan, funting.huang@gmail.com) and Michael Kenstowicz (Dept. of Linguist and Philosophy, Massachusetts Inst. of Technol., Cambridge, MA)

Unreleased stops, lacking a burst, have been claimed to have low perceptibility and are more likely to neutralize place contrasts. While this proposition has been supported by examining no-burst VC fragments spliced from released stops, little is known about the acoustic discriminability among true unreleased stops. This study fills this gap by analyzing the acoustic correlates of VC (where C = unreleased stops $p^-, t^-, k^-, \text{?}$) in Cantonese and Taiwanese Southern Min. Specifically, duration and formant transitions were estimated across three distinct contexts: VC-V vs. VC#V vs. VC#C. The preliminary results are (a) the formant transitions are effective cues to place contrasts of the unreleased stops: the labial has low $F2$ offset frequency, the coronal has high $F2$, and the dorsal has low $F3$, (b) the magnitude of transition cues varies with different contexts: the cues are more significant when followed by a vowel-initial, lexical morpheme (VC#V) than by a consonant-initial or functional morpheme (VC#C, VC-V), and (c) vowel raising is resisted in the dorsal-final environment. This finding may have implications for the phonotactic constraint $*[+high][high]$ in the two languages.

2aSC8. Ultrasound study for patterns of tongue movement in diphthongs, triphthongs, and vowel sequences in Korean, Japanese, and Mandarin. Boram Kim (Linguist, CUNY Graduate Ctr., The Graduate Ctr., City Univ. of New York, 365 Fifth Ave., New York, NY 10016, bkim@gradcenter.cuny.edu), Ai Mizoguchi (Speech-Language-Hearing Sci., Cu, New York, NY), and D. H. Whalen (Speech-Language-Hearing Sci., Cu, New Haven, CT)

Diphthongs have formant transitions sensitive to speaking rate, stress, vowel quality, and language ability. Tongue movement typically shows a functional “pivot” where the palate to tongue distance is almost constant; some combinations result in an “arch,” with only part of the tongue moving [Iskarous, J. Phon. 33, 363-381, 2005]. Here, ultrasound images of the tongue in Korean, Japanese, and Mandarin diphthongs and triphthongs (Mandarin) (vowel sequences in Korean and Japanese) were analyzed. Five repetitions of 6 nonwords were presented in Hangul, Hiragana or both Pinyin and a character for a word with the same vowel on tone 1. Diphthongs were [ai, ei, au] and triphthongs/sequence of three vowels were [iau, ieu, and uai]. All the ultrasound image frames for the target vowels were traced and superimposed. Preliminary results indicate that most vowel quality pairs resulted in a pivot pattern, with some arch patterns as well. Even in Mandarin triphthongs, there were generally two pivots, even though phonologically the sequence is considered a single vowel nucleus. Several approaches to quantifying this effect will be presented. It is possible that the success in producing a pivot could indicate mastery of production, both in development and in second language learning.

2aSC9. Exploring the acoustic characteristics of individual variation. Benjamin V. Tucker and Daniel Brenner (Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca)

Studies of the acoustic properties of words often analyze a small subset of words across a large population of speakers. Much of the previous research has not investigated the individual variation produced by a single speaker in large sets of words. The present study analyzes the individual variation produced by a male Western Canadian English speaker, who produced 26,800 English words and 9,600 pseudo-words. All pseudo-words were phonotactically licit and were generated using the software package Wuggy (Keuleers & Brysbaert, 2010). Each word has been force-aligned using the Penn Forced Aligner (Yuan & Liberman, 2008) and then hand corrected by trained phoneticians. We investigate the formant space, word pitch contours, segmental duration, and other acoustic characteristics relevant to classes of segments (such as center of gravity for fricatives). An acoustic comparison is performed between the words and pseudo-words. We explore the acoustic variation of the individual segments produced by this speaker and investigate his individual speech patterns. Finally, we consider the value of delving deeply into productions of a single speaker rather than relying on averaged summaries across a sample large sample.

2aSC10. Uncertainty of glottal airflow estimation during continuous speech using impedance-based inverse filtering of the neck-surface acceleration signal. Victor M. Espinoza (Dept. of Music and Sonology, Universidad de Chile, Compania 1264, 7th Fl., B Sector, Santiago 8340380, Chile, vespinoza@uchile.cl), Daryush Mehta, Jarrad Van Stan, Robert E. Hillman (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, Boston, MA), and Matías Zañartu (Dept. of Electron. Eng., Universidad Técnica Federico Santa María, Valparaíso, Valparaíso, Chile)

The aim of this work is to determine the uncertainty of non-invasive glottal aerodynamic measures that are obtained using subglottal impedance-based inverse filtering (IBIF) of the signal from a neck-placed accelerometer during continuous speech. Currently, we are studying the vocal behavior of individuals with typical voices and voice disorders by analyzing weeklong recordings using a smartphone-based ambulatory voice monitor. We extend on previously reported analyses of sustained vowel production using subglottal IBIF and move toward continuous speech applications where IBIF parameters are estimated in a frame-based approach. Selected voiced frames of both oral-airflow (baseline) and acceleration signal from the Rainbow Passage are used to build a probabilistic model of IBIF parameters to run

multiple random realizations of the inverse-filtered neck-surface acceleration signal. Confidence intervals are estimated for both the glottal waveform and derived features. The probabilistic model is tested using data from patients with vocal hyperfunction and matched-control subjects with normal voices at a comfortable pitch and loudness in an acoustically treated sound booth. Results show that model parameters follow approximately normal distributions, and the confidence intervals for the estimates of glottal aerodynamic measures were <10%, which is in close agreement with previously reported IBIF performance using sustained vowels.

2aSC11. Exertive modulation of coordinative structures in speech. Sam Tilsen (Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, tilsen@cornell.edu)

An articulatory study was conducted to investigate variability in the relative contributions of the upper lip, lower lip, and jaw to bilabial closure and opening tasks in speech. We currently do not know the extent to which the contributions of articulator subsystems may vary in the absence of linguistic contextual variation. One hypothesis is that variation in exertive mechanisms (e.g., arousal, effort, and attention) differentially affects articulator subsystems; this predicts that variation in articulator contributions will be nonstationary and will correlate with exertive variables. In this study, head movement during responses is considered a proxy for exertive variation. Nine experimental sessions were conducted in which six participants repeatedly produced the form [i.pa], instructed to do so as consistently as possible throughout the session. It was observed that distributions of relative articulator contributions differed substantially across participants and were non-stationary for all participants. Head movement during response production accounted for a substantial amount of variation in relative articulator contributions. These results show that interactions between subsystems in a coordinative structure are nonstationary and differentially susceptible to exertive modulations. This suggests that experimental manipulation of exertion can be used to investigate the organization of articulatory control.

2aSC12. Aeroacoustic consequences of tongue troughs in labiodentals. Christine H. Shadle (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu), Hosung Nam (English Lang. and Lit., Korea Univ., New Haven, CT), A. Katsika (Linguist, U.C. Santa Barbara, Santa Barbara, CA), Mark Tiede, and D. H. Whalen (Haskins Labs., New Haven, CT)

It has long been accepted that the main constriction for fricatives /f, v/ is formed by the lower lip pressing against the upper teeth, thus allowing the tongue to freely coarticulate with preceding and following segments. Here, electromagnetic articulometry data were obtained from 5 subjects in a study of tongue troughs, defined as a discontinuity in anticipatory coarticulation, such as when the tongue drops during a bilabial consonant in /i/ context. The corpus included /f/ and /v/ in VC(C)V contexts, where V = /i/ for C = /v/ , and V = {/i a u/} for C = /f/. The tongue moved down and back for /f/ in all vowel contexts; in /iC(C)i/ context, the troughs were deeper for long (VCCV) than short labial consonants, as predicted, and deeper for /f/ than for /p, b, v, m/, which was unexpected. There seems to be a secondary gesture specifying that the tongue be down during labiodental fricatives. Our hypothesis is that lowering the tongue for /f/ ensures that the airflow resistance will be due only to the labiodental constriction, thus providing an aeroacoustic advantage for the onset of turbulence. This is supported by comparison of asymmetric vowel contexts (e.g., /uffi/-/iffu/ tokens); tongue dorsum and blade sensors show the tongue moving quickly away from /i/ to /f/ position, in contrast to movement from /u/ to /f/. Aerodynamic considerations thus appear to be actively incorporated into the speech motor plan. [Work supported by NIH grant DC-002717.]

2aSC13. A comparison of lip positions for /i/ and /u/ in Bora. Steve Parker (Appl. Linguist, Graduate Inst. of Appl. Linguist, Dallas, TX) and Jeff Mielke (English, North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-8105, jimielke@ncsu.edu)

Bora is a Witotoan language spoken by about 750 persons in the Amazon jungle of Peru and 100 in Colombia. Its phonemic vowels are /i e a o i̯ u̯/ (Thiesen and Weber 2012). A contrast between a central and a back

vowel which are otherwise identical is theoretically significant since it shows that a binary feature [+/-back] is too weak to encode all phonological contrasts along the front/back dimension. The three high vowels of Bora have been acoustically confirmed with measurements of F1-F3 (Parker 2001), but there has been no articulatory investigation of these vowels. Impressionistically all of the Bora vowels except /o/ are articulated with unrounded lips. However, Ladefoged and Maddieson (1996) note that high back unrounded vowels in languages such as Japanese involve a gesture of lip compression or inrounding. Consequently, an important research question is whether the distinction between /i/ and /u/ in Bora can be relegated to a difference in lip positions rather than to a primary contrast in tongue backness? To test this hypothesis, we obtained video recordings of native speakers on location in a Bora village (6 males and 7 females), and we report lip position data from these recordings.

2aSC14. Gradient realization of Mandarin nasal codas. Yanyu Long (Linguist, Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, yl2535@cornell.edu)

Impressionistic studies suggested that Mandarin nasal codas optionally delete before vowels (/dan/+ai/->[dɑ.ai]) and assimilate to the following stops in place (/dan/+pai/->[dam.pai]). In this EMA study, we found that both processes are not phonological changes but are gradient processes modulated by speech rates. Three native Mandarin speakers read disyllabic words with n/ŋ codas before /a/ and /p/ in a carrier sentence at three speech rates. The tongue tip trajectories show that both nasals retain a reduced tongue gesture before /a/. The reduction INCREASES as speed INCREASES and is more variant in faster speech. The lower lip trajectories show that the labial gesture of /p/ occurs earlier when preceded by nasal codas. The time-normalized duration of this gestural advance DECREASES as speed INCREASES (and decreases at a slower rate in slower speech), suggesting less gestural overlap at faster speed. Such a relationship is opposite to the reduction-speed relationship. The articulatory evidence shows that although not perceptually salient, there is a reduced tongue gesture for nasals before vowels and gestural overlap between stops and preceding nasals. The non-linear effects of speed on gestural reduction and overlap suggest both are gradient phonetic processes; yet, their opposite relationships to speed suggest different implementation mechanisms.

2aSC15. Articulatory differences between glides and vowels. Dan Cameron Burgdorf and Sam Tilsen (Linguist, Cornell Univ., 28 Village Circle Apt. 2, Ithaca, NY 14850, dcb275@cornell.edu)

Glides bear similarities to both consonants and vowels, and align with different classes of phonological patterns in different languages. Limited prior studies have suggested that glides are realized with a greater degree of constriction and a shorter duration than vowels, but articulatory studies are rare and we do not know the relative importance of these properties. To determine how glides differ from vowels gesturally, an articulatory study was conducted with EMA. A two-dimensional stimuli continuum was constructed by manipulating the intensity and duration of high vowels (i, u) flanked by a low vowel (a), from a vowel-like extreme to a glide-like extreme, and participants were asked to imitate these stimuli immediately after hearing them. Results show interesting asymmetries and interactions; while intensity was not directly imitated through degree of constriction, it did affect intergestural timing, with low-intensity stimuli yielding shorter durations in production. The relationship between glides and vowels is more nuanced than any single featural difference.

2aSC16. Lemma frequency affects the duration of homographic noun/verb conversion “homophones.” Arne Lohmann (Dept. of English, Heinrich-Heine-Universität Düsseldorf, Universitätsstrasse 1, Duesseldorf 40225, Germany, arne.lohmann@hhu.de)

This paper reports empirical evidence for an effect of lemma frequency on the duration of homographic Noun-Verb homophones in spontaneous speech, e.g., *cut(N)* / *cut(V)*. In previous research on effects of lemma frequency, these words were not investigated due to a focus on heterographic homophones (e.g., Gahl 2008). However, testing the frequency hypothesis

on Noun/Verb homophones is of great theoretical relevance, as their representational status is especially controversial in both linguistic as well as psycholinguistic models of the mental lexicon. A mixed-effects analysis of speech data from the Buckeye corpus yields the result that the more frequent member of a Noun/Verb pair is pronounced with shorter duration, relative to its low-frequency twin. Generally speaking, this finding supports models of the mental lexicon in which entries are specified for syntactic category. Furthermore, this outcome is at odds with an account of “complete frequency inheritance” across homophones, as predicted by the Levelt production model. A separate analysis of the subsample of low-frequency words was carried out in order to further investigate possible frequency inheritance effects, as under an assumption of “partial inheritance.” No such effects were found. Taken together, the findings can be best accounted for in a model that assumes completely separate lexical representations for homophonous words.

2aSC17. Quantifying kinematic aspects of reduction in a contrasting rate production task. Mark Tiede (Haskins Labs., 300 George St., Ste. 900, New Haven, CT 06511, tiede@haskins.yale.edu), Carol Y. Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland, College Park, MD), Dolly Goldenberg (Linguist, Yale Univ., New Haven, CT), Vikramjit Mitra (Speech Technol. and Res. Lab., SRI Int., Menlo Park, CA), Hosung Nam (English Lang. and Lit., Korea Univ., New Haven, CT), and Ganesh Sivaraman (Elec. and Comput. Eng., Univ. of Maryland, Hyattsville, MD)

Electromagnetic articulometry (EMA) was used to record the 720 phonetically balanced Harvard sentences (IEEE, 1969) from multiple speakers at normal and fast production rates. Participants produced each sentence twice, first at their preferred “normal” speaking rate followed by a “fast” production (for a subset of the sentences two normal rate productions were elicited). They were instructed to produce the “fast” repetition as quickly as possible without making errors. EMA trajectories were obtained at 100 Hz from sensors placed on the tongue, lips, and mandible, corrected for head movement and aligned to the occlusal plane. Synchronized audio was recorded at 22050 Hz. Comparison of normal to fast acoustic durations for paired utterances showed a mean 67% length reduction, and assessed using Mermelstein’s method (1975), two fewer syllables on average. A comparison of inflections in vertical jaw movement between paired utterances showed an average of 2.3 fewer syllables. Cross-recurrence analysis of distance maps computed on paired sensor trajectories comparing corresponding normal:normal to normal:fast utterances showed systematically lower determinism and entropy for the cross-rate comparisons, indicating that rate effects on articulator trajectories are not uniform. Examples of rate-related differences in gestural overlap that might account for these differences in predictability will be presented. [Work supported by NSF.]

2aSC18. Physiological correlates of loud speech: Respiratory and intraoral pressure data. Laura L. Koenig (Adelphi Univ. and Haskins Labs., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu) and Susanne Fuchs (Leibniz Ctr. for General Linguist, Berlin, Germany)

Many previous studies have investigated how increased loudness affects speech production behavior, but authors have varied widely in the measures they have used, and few studies have systematically assessed relationships among respiratory, laryngeal, and acoustic measures. In this work, we present respiratory and aerodynamic (intraoral pressure) data on eleven German-speaking women who produced speech in regular and loud conditions in three different tasks: Reading short sentences, responding to questions, and producing spontaneous speech. Loudness variation was assessed naturally by varying speaker-experimenter distance. Respiratory behavior was assessed using inductance plethysmography, and intraoral pressure was obtained via a pressure transducer affixed to the hard palate. In the respiratory data, we measured inspiratory magnitude as well as the slope of the inspiratory and expiratory phases. In the intraoral pressure data, we searched automatically for the peak pressure value during pressives anterior to the transducer (viz., bilabials and alveolars). These physiological data will be related to previously-presented data on speech acoustics to begin to disentangle respiratory and supraglottal contributions to the characteristics of loud speech.

2aSC19. Phonation threshold pressure and the properties of the vocal tract. Lewis Fulcher (Dept. of Phys. and Astronomy, Bowling Green State Univ., Bowling Green, OH 43403, fulcher@bgsu.edu), Alexander Lodemeyer (Processmachinery and Systems Eng., Friedrich-Alexander Univ. Erlangen-Nuernberg, Erlangen, Bavaria, Germany), Stefan Kniesburges (Phoniatics and Pediatric Audiol., Univ. Hospital Erlangen, Erlangen, Germany), George Kaehler (Processmachinery and Systems Eng., Friedrich-Alexander Univ. Erlangen-Nuernberg, Erlangen, Bavaria, Germany), Michael Doellinger (Phoniatics and Pediatric Audiol., Univ. Hospital Erlangen, Erlangen, Germany), and Stefan Becker (Processmachinery and Systems Eng., Friedrich-Alexander Univ. Erlangen-Nuernberg, Erlangen, Bavaria, Germany)

In a classic 1988 paper, Titze presented arguments based on the dynamics of the motion of the air through the glottis and its relation to the pressures there to describe how the presence of the vocal tract should affect the phonation threshold pressure. He argued that the action of the intraglottal pressures due to the vocal tract and the motion of the vocal folds would be in phase, and thus the presence of the vocal tract should lower the threshold pressure by an amount that depends upon its inertence. Since the inertence of the vocal tract depends directly upon its length and inversely upon its cross sectional area, these arguments set the stage for quantitative studies of the connection of the geometry of the vocal tract with threshold pressure in both mathematical and physical models. To this end, two sets of experiments were carried out in Erlangen with a physical model of the vocal folds and a vocal tract whose dimensions could be varied. One set of experiments focused on the relationship of the threshold pressure and its frequency with the cross sectional area (areas varied from about 2 cm² to 12 cm²) and the other addressed the relationship of threshold pressure and its frequency with the length of the vocal tract (lengths varied in increments of 5 cm from about 4 cm to 54 cm). These measurements are compared with calculations done with the surface wave model and those done with a two-mass model.

2aSC20. Sensorimotor adaptation to auditory perturbation of speech is facilitated by noninvasive brain stimulation. Laura Haenchen, Ayoub Daliri, Sara C. Dougherty, Emily J. Thurston, Julia Chartrove, Tyler K. Perrachione, and Frank H. Guenther (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, haenchen@bu.edu)

Repeated exposure to disparity between the motor plan and auditory feedback during speech production results in a proportionate change in the motor system's response called auditory-motor adaptation. Artificially raising F1 in auditory feedback results in a concomitant decrease in F1 during speech production. Transcranial direct current stimulation (tDCS) can be used to alter neuronal excitability in focal areas of the brain. The present experiment explored the effect of noninvasive brain stimulation applied to the speech premotor cortex on the timing and magnitude of adaptation responses to artificially raised F1 in auditory feedback. Participants (N=18) completed a speaking task in which they read target words aloud. Participants' speech was processed to raise F1 by 30% and played back to them over headphones in real time. A within-subjects design compared acoustics of participants' speech while receiving anodal (active) tDCS stimulation versus sham (control) stimulation. Participants' speech showed an increasing magnitude of adaptation of F1 over time during anodal stimulation compared to sham. These results indicate that tDCS can affect behavioral response during auditory-motor adaptation, which may have translational implications for sensorimotor training in speech disorders.

2aSC21. Articulation and adaptation to altered auditory feedback. Sarah Bakst (Linguist, Univ. of California Berkeley, 1915 Bonita Ave., Studio A, Berkeley, CA 94704, bakst@berkeley.edu), John F. Houde (Otolaryngol., Univ. of California San Francisco, San Francisco, CA), and Keith Johnson (Linguist, Univ. of California Berkeley, Berkeley, CA)

Speakers listen to themselves while talking, and they use this auditory feedback to modify their speaking plans on-line [Houde and Jordan, *Science* 20;279(5354):1213-1216 (1998)]. This altered auditory feedback experiment uses ultrasound tongue imaging to investigate individual differences in adaptation under three conditions: (1) raising F1 in /ε/, (2) raising F2 in /o/, (3) raising F3 in /r/. Pilot data suggest that speakers may change both F1 and F2 in response to an altered F1, replicating Katseff *et al.* (2010, *JASA*

127(3), 1955). Principal components analysis of the ultrasound data reveals that these two acoustic changes are independently controlled. We will also test the role of individual differences in vocal tract morphology. Hard palate curvature affects variability in both articulation and acoustics [Brunner *et al.* (2009, *JASA* 125(6), 3936-3949)]: flatter palates have less acoustic stability (and greater flexibility) [Bakst & Johnson (2016, *JASA* 140(4), 3223-3223)], requiring greater articulatory precision to maintain acoustic consistency. We hypothesize that people with flatter palates will adapt to altered feedback faster and more completely because they (a) may have more detailed knowledge of their articulation-acoustics mapping and (b) have greater flexibility in their acoustic output.

2aSC22. Empirical eigenfunctions as a function of glottal adduction in excised hemilarynx experiments. David Berry (Surgery, UCLA, 31-24 Rehab, Los Angeles, CA 90095-1794, daberry@ucla.edu)

For three excised human male hemilarynxes, vocal fleshpoints and empirical eigenfunctions were computed along the vocal fold surface. For two larynges, an increase in adduction resulted in an increase in lateral and vertical oscillation amplitudes and an improved energy transfer from the airflow to the vocal fold tissues. In contrast, the third larynx exhibited a decrease in oscillation amplitudes. By evaluating the empirical eigenfunctions, this decrease in oscillation amplitudes was associated with an unbalanced oscillation pattern with predominantly lateral amplitudes. These results suggest that adduction facilitates the phonatory process by increasing vibrational amplitudes. However, this relationship holds only when a balanced ratio between the vertical and lateral displacements is maintained. Indeed, it appears that a balanced vertical-lateral oscillation pattern may be more beneficial to phonation than strong periodicity with predominantly lateral vibrations.

2aSC23. Non-linear dimensionality reduction for correlated tongue measurement points. Jaekoo Kang (Speech-Language-Hearing Sci. Program, CUNY Graduate Ctr., 3547 34th St., Apt. 1E, Long Island City, NY 11106, jkang@gradcenter.cuny.edu), D. H. Whalen (Speech-Language-Hearing Sci. program, CUNY Graduate Ctr., New York, NY), and Hosung Nam (Haskins Labs., New Haven, CT)

The tongue surface is a good indicator of the main supralaryngeal articulation of speech, and quantifying it with more points to measure increases accuracy. However, unlike acoustic variables (e.g., formants), articulatory variables (e.g., flesh-point pellets or multiple measurement points on an ultrasound image) are highly correlated to one another. Projecting the correlated variables with high dimensionality to lower is necessary. This study employs a nonlinear reduction method, Autoencoder (Hinton & Salakhutdinov, 2006), and compares its performance to that of Principal Component Analysis, which assumes orthogonality and linearity of dimensions. The two methods were applied to eight English vowels in clear speech from Wisconsin X-ray Microbeam dataset (Westbury, 1990). Root-mean-squared errors measured after the data reconstruction were analyzed by vowel type and pellet locations. Preliminary results with one speaker exhibited a slightly better performance than the nonlinear method, especially for some vowels (/i,u/). More speakers and time frames are predicted to lead to larger improvements of the nonlinearity analysis over the linear PCA. Similar tests of the two methods will be performed on the variabilities of the corresponding acoustic data. It is predicted that nonlinear analyses will make the variabilities in the acoustic and articulatory domains more comparable than previously assumed.

2aSC24. Sensitivity and specificity of auditory feedback driven articulatory learning in virtual speech. Jeffrey J. Berry, Ramie Bagin (Speech Pathol. & Audiol., Marquette Univ., P.O. Box 1881, Milwaukee, WI 53201-1881, jeffrey.berry@marquette.edu), James Schroeder (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 and acoustic data characterizing how the form of synthesized auditory feedback in virtual speech affects the sensitivity and specificity of articulatory learning. The term "virtual speech" refers to talker-manipulated synthesized speech controlled

in real time using electromagnetic articulography (EMA). In the current work, 36 participants (4 with dysarthria) participated in a learning experiment requiring them to control an articulatory speech synthesizer using movements of the tongue, lips, and jaw. Participants were divided among two experimental conditions: (1) an “unmatched” condition, during which all participants received auditory feedback based on common articulatory synthesis settings (neither formant working space nor fundamental frequency were distinguishable between talkers); and (2) a “matched” condition, during which the articulatory synthesis parameters were adjusted to mimic the formant working space and average fundamental frequency of the learner. Analyses focused on the kinematic and acoustic differences in learning between the two conditions. Results suggest that the sensitivity and specificity of articulatory learning is affected by the extent to which the auditory feedback matches the learner’s familiar acoustic working space. Findings have implications regarding how the acoustic characteristics of perceived speech affect sensorimotor integration and learning in typical talkers and individuals with dysarthria.

2aSC25. Articulatory reuse in “good-enough” speech production strategies. Matthew Faytak (Linguist, Univ. of California, Berkeley, 2632 San Pablo Ave. Apt. A, Berkeley, CA 94702, mf@berkeley.edu)

Given a novel speech motor task, a speaker may optimize execution of the task for accuracy by taking feedback into account, or revert to more habitually used productions which provide a precise output not entirely optimized for accuracy. An articulatory examination comparing L1 and L2 productions was carried out in part to assess the roles of optimization and reversion to habit for individual speakers. Native speakers of American English learning French, ($n=30$, 9 males) with a variety of levels of exposure to the L2, were recorded producing the monophthongs of both languages and two English approximants (/r/ and /l/) using ultrasound tongue imaging, video of lip shape, and audio. Principal component analyses run on tongue ultrasound and lip shapes of individual speakers reveal that production of L2 French essentially within L1 English articulatory habit is typical; in several cases, the approximants /r/ and /l/ are essentially reused as French vowels (e.g., /ø/ and /u/). However, a slight optimization toward native-like productions can be observed in speakers with longer exposure to the L2.

2aSC26. Amplitude envelope kinematics of speech: Parameter extraction and applications. Lei He and Volker Dellwo (Phonet. Lab, Univ. of Zurich, Dübendorfstrasse 32, Zurich 8051, Switzerland, lei.he@uzh.ch)

We model the amplitude envelope of a speech signal as a kinematic system and calculate its basic parameters: displacement, velocity, and acceleration. Such system captures the smoothed amplitude fluctuation pattern over time, illustrating how energy is distributed across the signal. Although the pulmonic air pressure is the primary energy source of speech, the amplitude modulation pattern is largely determined by articulatory behaviors, especially mandible and lip movements. Therefore, there should be a correspondence between signal envelope kinematics and articulator kinematics. Previous research showed that a tremendous amount of speaker idiosyncrasies in articulation existed. Such idiosyncrasies should therefore be reflected in the envelope kinematics as well. From the signal envelope kinematics, it may be possible to infer individual articulatory behaviors. This is particularly useful for forensic phoneticians who usually have no access to articulatory data, and clinical speech pathologists who usually find it difficult to make articulatory measurements in clinical consultations.

2aSC27. Are idiosyncrasies in vowel production free or learned? A study of variants of the French vowel system in biological brothers. Lucile rapin (Linguist, Université du PQ a Montreal, Montreal, QC, Canada), Jean-Luc Schwartz (GIPSA-Lab, Grenoble, France), and Lucie Menard (Linguist, Université du PQ a Montreal, CP 8888, succ. Centre-Ville, Montreal, QC H3C 3P8, Canada, menard.lucie@uqam.ca)

Speech production displays a number of idiosyncrasies that are individual variations in the way speakers achieve phonetic contrasts in their language. It was shown previously [Ménard L. et al., *Speech Commun.* 2008, 50(1), 14-28] that idiosyncrasies in the production of the height contrast in oral vowels in French are characterized by large variations in the distribution of F1 values,

which are associated with a stability of F1 for a given height degree, independent of the place of articulation (front vs. back) and rounding. The current study aimed to assess whether these idiosyncrasies are random or induced by the learning environment. Ten pairs of French Canadian adult male siblings were recruited, and each individual was asked to produce ten repetitions of the ten French oral vowels, which were recorded. F1 values were extracted using linear predictive coding algorithms. There was a trend towards imposed variations, since the distances between F1 values for brothers for a given vowel were significantly smaller than the corresponding distances between speakers who were not brothers. However, the F1 values within pairs of brothers were significantly correlated for only two of the six mid-high or mid-low vowels. Thus, it appears that a large part of the idiosyncrasies in the pronunciation of vowels were random and differed between brothers.

2aSC28. Random effects and the evaluation of the uniform scaling hypothesis for vowels. Terrance M. Nearey (Linguist, Univ. of AB, 4-32 Assiniboia Hall, University of AB, Edmonton, AB T6G 0A2, Canada, tnearey@ualberta.ca), Santiago Barreda (Linguist, Univ. of California, Davis, CA), Michael Kieft (School of Commun. Disord., Dalhousie Univ., Halifax, NS, Canada), and Peter F. Assmann (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, Richardson, TX)

The uniform scaling hypothesis suggests that formant frequencies of vowels of one speaker can accurately predict those of any other speaker of the same dialect by applying a single multiplicative scale factor. Fant (*STL QPSR*, 2-3, 1-19, 1975) questioned this assumption and presented graphical evidence that scale factors vary between adult men and women for different vowels in ways that are systematically related to their position in the vowel space. However, the issue is complicated by the fact that average vowel datasets may contain multiple sources of variation, including dialect mixture. Statistical evaluation of the uniform scaling hypothesis (or any systematic deviations from it) need to account for multiple sources of variation. In preliminary random-effects analyses of formant data produced by individual speakers from three geographically distinct dialect regions of American English, we found that while relatively modest systematic trends in nonuniformity may exist, their magnitude may be smaller than suggested by earlier work and their assessment may be strongly influenced by other sources of variation within geographical dialect regions. We will present extensions of our preliminary analyses that will include speech data from Texas, Alberta, and Nova Scotia collected in our laboratories.

2aSC29. Speech acoustics can be modulated by cognitive interference in a vowel-modified Stroop task. Caroline A. Niziolek (Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., SLHS, Rm. 351, Boston, MA 02135, carrien@bu.edu), Kimberly R. Lin (Neurosci., Boston Univ., Boston, MA), Sara D. Beach (Speech and Hearing BioSci. and Technol., Harvard Med. School, Boston, MA), Ian A. Quillen (Neurosci., Boston Univ., Boston, MA), and Swathi Kiran (Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

How are speech acoustics influenced by cognitive processes? In the current study, we used a novel variant of the Stroop test to measure whether the interference between color naming and reading could modulate vowel formant frequencies. Seventeen healthy participants named the color of words in three categories: (1) congruent words (e.g., “red” written in red), (2) color-incongruent words (e.g., “green” written in red), and (3) vowel-incongruent words with phonetic properties that partially matched their color (e.g., “rid” written in red). We hypothesized that the cognitive effort needed to inhibit reading—saying “red,” not “rid”—could affect vowel acoustics. For example, the correct spoken response (“red”) could more acoustically resemble the inhibited word “rid;” alternatively, the vowel could be influenced in the opposite direction, resembling “rad,” which would serve to accentuate the acoustic contrast between the spoken and inhibited words. As expected, participants were slower to produce words on color-incongruent trials than on congruent trials. Interestingly, vowel-incongruent trials were not significantly slower than congruent trials, but preliminary acoustic analyses of the first formant (F1) showed that some subjects systematically modulated their productions in the presence of incongruent vowels. This finding lends insight into how the brain integrates multiple pieces of information to produce speech.

2aSC30. Closed-syllable vowel laxing: A contrast enhancement strategy. Benjamin Storme (Linguist and Philosophy, MIT, 16 Wilson Ave., Somerville, MA 02145, bstorme@mit.edu)

Closed-syllable vowel laxing describes the cross-linguistic tendency for high and mid vowels to have higher F1 values and more central F2 values in closed than in open syllables. This pattern is often analyzed as resulting from vowel shortening in closed syllables. However, vowel undershoot does not generally result in an increase of F1 for mid vowels. This paper tests an alternative hypothesis according to which laxing is a strategy to enhance coda-consonant place contrasts, with lower and more central vowels providing more informative closure transitions than higher and more peripheral vowels. Two native French speakers were recorded uttering C1VC2 nonce words with C1, C2={p,t,k} and V={i,y,u,e,ø,o,ε,æ,ɔ,a}. 85 English and French hearers were presented with the stimuli without word-final bursts and were asked to identify the place of the word-final consonant. In accordance with the enhancement hypothesis, lowering accompanied by centralizing was found to improve [p]-[k] and [t]-[k] contrasts for front unrounded vowels and [p]-[k], [t]-[k], and [p]-[t] contrasts for back vowels. These contrasts were not systematically more distinct after front rounded vowels than after back and unrounded front vowels with similar F1 values (e.g., [y] vs. [i]/[u]), suggesting that centralizing alone is not sufficient to enhance place contrasts.

2aSC31. Quantifying sonority contour: A case study from American English. Suyeon Yun (Ctr. for French and Linguist, Univ. of Toronto Scarborough, 1265 Military Trail, Humanities Wing, H427, Toronto, ON M1C 1A4, Canada, suyeon.yun@utoronto.ca)

Previous studies have argued that the most reliable phonetic correlate of sonority is intensity (e.g., Parker 2002, 2008, Jany *et al.* 2007). However, those studies have only considered intensity of a single segment. This paper investigates the phonetic correlate of sonority contour in consonant clusters. 10 native speakers of American English (5 male, 5 female) read 33 monosyllabic English words that begin with a bi- or tri-consonantal cluster (e.g., *play*, *stray*) embedded in a frame sentence (“Father saw ‘___’ again,” used in Parker 2008). Measured first were (i) an average RMS and (ii) sound level minima of each consonant in the cluster C1C2, and the sonority contour was quantified by subtracting the intensity value of C1 from the intensity value of C2. Also, (iii) actual intensity slopes in the transition between the two consonants were measured. Results show that the intensity contours calculated based on (i) and (ii) do not always correspond to the intensity slopes (iii), while both of them are in general correlated with the sonority contour. It will also be suggested that it is intensity slopes (iii) that play a crucial role in consonant cluster perception and in phonological phenomena involving consonant clusters.

2aSC32. Phoneme distribution and phonological processes of orthographic and pronounced phrasal words by syllable structure in the Seoul Corpus. Byunggon Yang (English Education, Pusan National Univ., 30 Changjundong Keumjunggu, Pusan 609-735, South Korea, bygyang@pusan.ac.kr)

This study examined the phoneme distribution and phonological processes of orthographic and pronounced phrasal words according to syllable structure in the Seoul Corpus of spontaneous speech produced by 40 Korean speakers. To achieve the goal, the phrasal words were extracted from the transcribed label scripts of the Seoul Corpus using Praat. Then, the onsets, peaks, codas, and syllable types of the phrasal words were analyzed using an R script. Results revealed that *k0* was most frequently used as an onset in both orthographic and pronounced phrasal words. Also, *aa* was the most favored vowel in the Korean syllable peak with fewer phonological processes in its pronounced form. For the codas, *nn* accounted for 34.4% of the total pronounced phrasal words and was the varied form. From syllable type classification of the Corpus, CV appeared to be the most frequent type followed by CVC, V, and VC from the orthographic forms. Overall, the onsets were prevalent in the pronunciation more than the codas. From the results, the author concludes that an analysis of phoneme distribution and phonological processes in light of syllable structure can contribute greatly to the understanding of the phonology of spoken Korean.

2aSC33. A landmark-based approach to transcribing systematic variation in the implementation of /t, d/ flapping in American English. Suyeon Yun (Ctr. for French and Linguist, Univ. of Toronto Scarborough, 1265 Military Trail, Humanities Wing, H427, Toronto, ON M1C 1A4, Canada, suyeon.yun@utoronto.ca), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA)

A model of human speech processing based on individual cues to distinctive features of phonemes, such as the acoustic landmarks (abrupt spectral changes) that signal manner features, is proposed to provide a more accurate account of American English flapping of /t/ and /d/ than an allophonic or phone-based model. To test this hypothesis, this study analyses the phonetic realization of /t, d/ in the context of flapping using the acoustic landmark cues of abrupt stop closure, abrupt stop release and glide-like amplitude minimum (Stevens 2002), in subsets of the TIMIT corpus. Results show that the majority of flapped variants of /t, d/ preserve their stop closure landmark and there are several cases where they preserve both of their stop landmarks (stop closure and stop release), while exhibiting the landmark modification for flapping (e.g., t-glide-+). Additionally, flapped /t/ is more likely to maintain stop landmarks than flapped /d/. This is unexpected from the traditional view of the flap as a categorical phenomenon, and suggests that acoustic landmarks are useful in capturing systematic phonetic variation in flapping. It will be important to test whether this landmark-based analysis yields a better result in automatic speech recognition than (allo)phone-based approaches.

2aSC34. Detecting glides and their place of articulation using speech-related measurements in a feature-cue-based model. Adrian Y. Cho (Harvard-MIT Program in Speech and Hearing BioSci. and Technol., 50 Vassar St., Rm. 56, Speech Commun. Group, Cambridge, MA 02139, aycho@g.harvard.edu), Anita Y. Liu (Speech Commun. Group, Res. Lab. of Electronics, MIT, Quincy, MA), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Speech Commun. Group, Res. Lab. of Electronics, MIT, Cambridge, MA)

An algorithm was developed for detecting glides (/w/, /j/, /r/, /l/, or /h/) in spoken English and detecting their place of articulation using an analysis of acoustic landmarks [Stevens 2002]. The system uses Gaussian mixture models (GMMs) trained on a subset of the TIMIT speech database annotated with acoustic landmarks. To characterize the glide tokens extracted from the speech samples, the following speech-related measurements were calculated: energy in four spectral bands (E1-E4), formant frequencies (F1-F4), and the time derivatives of E1-E4 (E1'-E4'); the fundamental frequency (F0) and magnitude difference of harmonics (H1-H2, H1-H4) were also included. GMMs were then trained on a subset of the tokens to learn the characteristics of each category for two distinct tasks: distinguishing glide landmarks from the set of all landmark types (identification task), and determining the place of articulation given a glide landmark (categorization task). The classifier used the maximum posterior probability of a speech sample conditioned on each of the trained GMMs. The performance of the algorithm was evaluated with median F-scores, and results suggest that the measurements at acoustic landmarks provide salient cues to glide detection and categorization.

2aSC35. A new metric for calculating acoustic dispersion in stop inventories. Ivy Hauser (Linguist, Univ. of Massachusetts Amherst, 650 North Pleasant St., Amherst, MA 01060, ihauser@linguist.umass.edu)

Dispersion Theory (DT; Liljencrants and Lindblom, 1972) claims that acoustically dispersed vowel inventories should be typologically common. Literature on DT has focused on vowels, where the predictions are robust, and less work has been done on consonants. This paper uses vocal tract model data of stops (as in Schwartz *et al.* 2012) to extend the predictions of DT to consonants, revealing problems with the conventional method of calculating dispersion. Dispersion is often quantified using triangle area between three category means as points in acoustic space. This approach ignores distributions and reduces entire acoustic categories (which have large variances and different distribution shapes) to single points. Within-category variance is a factor in DT (Lindblom, 1986) and experimental data shows that it affects perception (Clayards 2008), yet conventional dispersion metrics do not take it into account. Here, a new metric based on the Jeffries-

Matusita distance is proposed and compared with the more conventional mean to mean distance approach. The incorporation of covariance better reflects human perception, which has implications for considering dispersion in any acoustic space. Nevertheless, this does not recover the predictions of DT, suggesting DT does not apply to consonants and vowels in the same way.

2aSC36. Toward an analysis of Spanish glides in the acoustic landmark framework. Violet Kozloff (Wellesley College, Unit 6018 21 Wellesley College Rd., Wellesley, MA 02481, vkozloff@wellesley.edu), Stefanie Shattuck-Hufnagel (MIT, Boston, MA), and Jeung-Yoon Choi (MIT, Cambridge, MA)

Stevens (2002) proposes that the distinctive feature [glide] is signaled by an acoustic landmark, i.e., an amplitude/F1 minimum, usually during a phoned region, but this hypothesis has not been tested extensively in languages other than American English. This study analyzes acoustic realizations of tapped /r/ and trilled /r/ sounds in European Spanish, identifying a range of acoustic realizations for these consonants, including glides, and proposing criteria for identifying Spanish tap and trill landmarks in the speech signal. The speech sample of 200 tokens was drawn from the Albayzin corpus, which includes recordings of read Castilian Spanish from male and female speakers. Tokens were characterized by the number of amplitude minima (or occlusions) as well as the presence or absence of vocal fold vibration and noise. Additional factors analyzed include the rate of amplitude modification (tongue tip vibration), and contextual factors, including word and syllable position, stress, and consonant clusters. These moments of abrupt change (amplitude inflection points) provide cues to the manner features of the speaker's intended words, and are hypothesized to play a significant role in perceptual processing and word recognition. These initial results for /r/ provide the basis for extension of this analysis to other Spanish glides.

2aSC37. Similarity measurement based rest position re-initialization in the MRI vocal tract image sequences. Xi Liu (ESPCI, 10 Rue Vauquelin, ESPCI Paris, Paris 75005, France, 1992xi.liu@gmail.com) and Kele Xu (College of Electron. Sci. and Eng., National Univ. of Defense Technol., Paris, France)

Magnetic resonance imaging is often used for the speech production research. One important aspect is to segment the vocal tract in the image sequence. However, during the deformation of the vocal tract, it is highly possible that the landmark deviates from its correct position and is in need of reinitialization. During the natural speech production of the subject, the vocal tract may return to its initial position during the pause at the end of a sentence. This could be used to reset the landmark to its correct position. In order to determine the pause in the image sequence, we used similarity based measurements to compare the similarity between the current frame and the first frame, which is the beginning of a sentence and the vocal tract is at the rest position. These measurements include Structural Similarity (SSIM), Complex Wavelet Structural Similarity (CW-SSIM), Visual Information Fidelity in Pixel (VIFP), Peak Noise to Signal Ratio (PSNR), etc. We found CW-SSIM outperformed the other methods. We calculated the similarity measurements and they varied periodically during the speech. CW-SSIM returned to a maximum of around 0.9, which indicated that the vocal tract returned to its initial position. The rest of the similarity measurements returned to a maximum value greatly deviated from 1, which indicated that the CW-SSIM was the best candidate.

2aSC38. Uncontrolled manifold method to speech production. Hosung Nam (Dept. of English Lang. and Lit., Korea Univ., 145 Anam-ro, Seongbuk-gu, Seoul 02841, South Korea, hnam@korea.ac.kr), Jaekoo Kang (Speech-Language-Hearing Sci. program, CUNY Graduate Ctr., Long Island City, NY), and Elliot Saltzman (Dept. of Physical Therapy and Athletic Training, Boston Univ., Boston, MA)

Speech production is a highly skilled sensorimotor activity defined by articulatory or acoustic coordinates. To compare the variabilities of those two conceptualizations, issues of dimension reduction, normalization, incompleteness of information, etc., need to be taken into account. Uncontrolled manifold (UCM) method analyzes high-dimensional movement data-set with respect to the outcomes that count as successful tasks. It divides the variability in the data into two parts: "bad" variability associated with motion within the controlled manifold (CM) that would lead to an error in the task and "good" variability within the uncontrolled manifold (UCM) that do not harm to the task. The smaller ratio indicates both tighter control (less variability in the CM) and greater flexibility (more variability in the UCM). The UCM method is applied to the Wisconsin X-ray microbeam data. We first constructed a neural-net-based forward mapping from articulators to acoustics. The inter-layer weight matrices and the outputs of each layer in the trained forward model are then used to compute the elements of this forward model's Jacobian matrices; the Jacobians are then used to compute σ_{CM}/σ_{UCM} ratios. We further compare these ratios across data obtained in various linguistic conditions.

2aSC39. Anatomically oriented Principal Components Analysis of three-dimensional tongue surfaces. Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., Bloomington, IN), Max Nelson, Kenneth de Jong, and Kelly Berkson (Linguist, Indiana Univ., 1021 East 3rd St - Memorial Hall 322 E, Bloomington, IN 47405, maxnelso@umail.iu.edu)

A procedure for carrying out Principal Components Analyses of three-dimensional tongue surfaces segmented from 3D/4D volumetric ultrasound images is presented. The segmented surface is transformed to a spherical coordinate system with the origin defined at the anterior visible extreme of the tendon of the genioglossus (near the mandibular symphysis). Principal Components Analyses of tongue surface shapes for monosyllabic real words carried out in this spherical coordinate system are robust to variations in the location of the origin, and show similarities across speakers, based on a corpus of 10 speakers.

2aSC40. High-resolution speech directivity measurements. Claire Pincock, Timothy W. Leishman, and Jenny Whiting (Brigham Young Univ., 159 E 300 S #3, Provo, UT 84606, mckellar.claire@gmail.com)

A measurement system has been developed at Brigham Young University to assess high-resolution directivity data produced by human subjects. The system incorporates 2522 unique sampling positions over a sphere and has been used to acquire directivity data of several female and male talkers repeating phonetically balanced passages. Both polar and balloon plots of these data have been generated, along with similar plots corresponding to gender-specific averages and spherical-harmonic expansions. The results will be used for speech radiation studies and architectural acoustics simulations. This presentation reports the results and compares the directivity averages for both genders.

Session 2aSPa

Signal Processing in Acoustics: Topological Signal Processing

Jason E. Summers, Chair

Applied Research in Acoustics LLC, 1222 4th Street SW, Washington, DC 20024-2302

Chair's Introduction—9:15

Invited Papers

9:20

2aSPa1. Topological features in signal processing using frame theory and persistent homology. Mijail Guillemard (Mathematics, TU Hamburg, Hamburg Univ. of Technol. Inst. of Mathematics (E-10), Hamburg, Hamburg 21073, Germany, mguillemard@gmail.com)

We present some interactions between frame theory and persistent homology as a new way to construct classification mechanisms in signal processing. On the one hand, frame theory generalizes basic ideas from time-frequency analysis including aspects of short term Fourier transformations and wavelet theory. On the other hand, persistent homology provides new algorithms applying concepts from algebraic topology to data analysis. The question of finding adequate sparse representations of data can be seen from several points of view, including dimensionality reduction and modern developments in neural networks. Persistent homology, as a topic in topological data analysis, presents alternative mechanisms for finding adequate sparse representations of data. We explain some interactions between these tools with applications to the analysis of acoustic signals.

9:40

2aSPa2. The performance of topological classifiers on sonar data. Michael Robinson (Mathematics and Statistics, American Univ., 4400 Massachusetts Ave. NW, 226 Gray Hall, Washington, DC 20016, michaelr@american.edu)

For various reasons, synthetic aperture sonar (SAS) target classification in various clutter contexts is usually done using a data-driven, machine learning approach. Unfortunately, the resulting feature set can be rather inscrutable—what features is it really using? Topological methods are particularly well-aligned with the goal of gaining insight into physical processes, since they highlight symmetries which are driven by these physical processes. For instance, collating multiple image looks of a round object uncovers rotational symmetries in an appropriate feature space derived from the images. The use of topological invariants allows one to infer that the object is round by reasoning about its feature space. The fact that sonar target signatures are (mostly) translation invariant in range can also be deduced from topological invariants. I will describe a principled, foundational analysis of target echo structure through the lens of topological signal processing, and then analyze the performance of this approach as compared to more traditional classification methods.

10:00

2aSPa3. Sliding windows and persistence. Jose Perea (Computational Mathematics, Sci. and Eng., Michigan State Univ., 1501 Eng. Bldg., East Lansing, MI 48824, joperea@msu.edu) and Chris Traile (Elec. and Comput. Eng., Duke Univ., Durham, NC)

The use of geometric and topological ideas as a means to tackle problems in signal analysis has seen a sharp increase in the last few years. The goal of this talk is to show how ideas from dynamical systems (e.g., time delay embeddings) with tools from topological data analysis (e.g., persistent homology) allows one to extract highly non-trivial features from vector-valued time series data. As an example, we describe a paradigm for quantifying (quasi)periodicity in video data and provide applications including the study of gene regulatory networks in biology, biphonation in mammals, and speech pathologies in humans.

10:20–10:40 Panel Discussion

Session 2aSPb**Signal Processing in Acoustics, Engineering Acoustics, and Architectural Acoustics: Signal Processing for Directional Sensors I**

Kainam T. Wong, Chair

*Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, DE 605, Hung Hom KLN, Hong Kong****Invited Paper*****11:00**

2aSPb1. On the general connection between wave impedance and sound intensity. Domenico Stanzial (Res. Section of Ferrara, CNR - Inst. of Acoust. and Sensors "Corbino," v. Saragat, 1, c/o Phys. and Earth Sci. Dept., Ferrara 44122, Italy, domenico.stanzial@cnr.it) and Carlos E. Graffigna (Int. Doctorate Program, Universidad Nacional de Chilecito - Univ. of Ferrara - CNR-IDASC, Ferrara, Italy)

This paper presents the generalization to non-monochromatic non-monodimensional fields of the equation linking the complex sound intensity to the wave impedance/admittance already introduced with a different form in [D. Stanzial and C. E. Graffigna "On the connection between wave impedance, sound intensity and kinetic energy in monochromatic fields," accepted for publication on POMA, Dec. 23, 2016]. Computer simulations have been now carried out, both for wave impedance and admittance, in quasi-stationary bi-dimensional waves fields with different reflection coefficients and spectral compositions. It turns out that the equation is validated in all space points of the sound field for each spectral component. This allows primarily to calculate the active intensity vector field as the vector sum of all vector fields obtained as the spectral components of the active intensity and therefore to determine the reactive intensity magnitude by simply subtracting the modulus of the so obtained active intensity from scalar field of the apparent intensity. This grand result will allow to develop a precision device for measuring 3D sound intensity and its active and reactive spectral components.

Contributed Paper**11:20**

2aSPb2. Bias error comparison for plane-wave acoustic intensity using cross-spectral and phase-and-amplitude-gradient-estimator methods. Daxton Hawks (Brigham Young Univ. - Idaho, Rexburg, ID), Tracianne B. Neilsen, Kent L. Gee, and Scott D. Sommerfeldt (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu)

Acoustic vector intensity relies on the product of the acoustic pressure and particle velocity. The particle velocity is typically approximated via Euler's equation using the gradient of the complex pressure across closely spaced microphones, which is traditionally found using the cross-spectral density. In contrast, the phase-and-amplitude-gradient-estimator (PAGE)

method [Thomas et al., *J. Acoust. Soc. Am.*, **137**, 3366-3376 (2015)] relies on gradients of pressure magnitude and phase. For a broadband source this allows for the phase to be unwrapped, which extends the usable bandwidth of the intensity calculation well above the spatial Nyquist frequency. The benefits of the PAGE method are evident in plane wave tube measurements in which microphones spaced 90 cm apart yield accurate intensity values at frequencies at least ten times the spatial Nyquist frequency. This represents an increase in bandwidth of 30 times over the traditional method. The bias errors for the traditional method for calculating acoustic intensity are reviewed and compared with the bias errors for the PAGE method for the case of both two and three microphone intensity probes in a plane-wave tube environment. [Work supported by the National Science Foundation.]

Invited Papers**11:40**

2aSPb3. Precision device for measuring the three dimensional spectral intensity. Domenico Stanzial (Res. Section of Ferrara, CNR - Inst. of Acoust. and Sensors "Corbino," Ferrara, Italy) and Carlos E. Graffigna (Int. Doctorate Program, Universidad Nacional de Chilecito - Univ. of Ferrara - CNR-IDASC, v. Saragat 1, c/o Phys. and Earth Sci. Dept., Ferrara I-44122, Italy, carlos.graffigna@idasc.cnr.it)

On the basis of processing algorithms developed by the same authors in a companion paper ["On the general connection between wave impedance and sound intensity", 173rd Meeting of the Acoustical Society of America and the 8th Forum Acusticum, Boston MA, 25-29 June 2017] where the equation between the complex sound intensity and the specific acoustic impedance/admittance has been stated and numerically validated in general form, the block diagram of a possible device for precision measurement of 3D spectral intensity is proposed here. In order to test its functionality, some measurements have been carried out, inside a tube of 28x28 cm² square

section and 4 m long, which was terminated with panels of different materials. Measurements have been carried out at different positions along the tube's axis by means of a 3D pressure-velocity probe below and above the cutoff frequency of the tube so to include also the effects of transversal modes. Preliminary results of such measurements are reported and discussed briefly here.

12:00

2aSPb4. Source localization with three-dimensional sound intensity probe with high precision. Jeong-Guon Ih, In-Jee Jung, and Jung-Han Woo (Mech. Eng., KAIST, 373-1 Guseong-Dong, Yuseong-Gu, Daejeon 305-701, South Korea, J.G.Ih@kaist.ac.kr)

When an array module measuring three-dimensional sound intensity is employed for detecting the sound source, a compact space usage and small number of sensors are advantageous than the other source localization methods. However, because of severe bias errors, it has not been popular. We analyze the major sources of bias estimation error and seek for the compensation method. Spectral bias error is due to the reflected signal from the environment, which is proportional to the difference of distance between direct and reflective paths. Spatial bias error is due to the inhomogeneous directivity of the intensity module stemming from discrete arrangement of sensors on the hypothetical sphere surrounding sensors. Simulation with changing the source direction by 1 deg. in spherical angle can generate an error map for all incidence angles. A measurement is conducted using a tetrahedral intensity module with 30 mm spacing for the compensation of errors. Low pass filtering of the cross spectral density function is used for the spectral bias error, and spherical error map is used for the directional bias error. By compensating such bias errors, it is shown that the localization errors of all bearing angles are less than 1° in an anechoic chamber when $kd < 1.1$.

MONDAY MORNING, 26 JUNE 2017

ROOM 306, 9:15 A.M. TO 12:20 P.M.

2a MON. AM

Session 2aUWa

Underwater Acoustics: Sound Propagation and Scattering in Three-Dimensional Environments I

Ying-Tsong Lin, Cochair

Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543

Frédéric Sturm, Cochair

Acoustics, LMFA, Centre Acoustique, Ecole Centrale de Lyon, 36, avenue Guy de Collongue, Ecully 69134, France

Chair's Introduction—9:15

Invited Papers

9:20

2aUWa1. Gaussian beam tracing for calculating the broadband field in three-dimensional environments. Michael B. Porter and Laurel Henderson (HLS Res., 3366 N. Torrey Pines Ct., Ste. 310, La Jolla, CA 92037, mikeporter@hlsresearch.com)

Ray tracing methods have a long history in underwater acoustics going back to a paper by H. Lichte in 1919 which also predicted the SOFAR channel. They remain extremely valuable today, partly because they present an intuitive view of sound propagation that readily allows for many extensions. For instance, targets and boundaries with complicated scattering can be included in a natural way; similarly, motion of boundaries, sources, receivers, and the ocean itself are easily treated. This talk will focus on the 3D extension with particular emphasis on broadband waveforms such as chirps or waveforms due to acoustic modems.

9:40

2aUWa2. 3-D ocean acoustics with normal modes. David P. Knobles (Knobles Sci. and Anal., PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com)

A new derivation is presented for coupled mode equations applicable to 3-D ocean environments possessing strong horizontal variability. The 3-D acoustic field is represented by a bi-orthonormal expansion with both the depth- and azimuthal-dependent eigenfunctions. Two classes of coupling integrals emerge. One class is associated with azimuthal modes and their azimuthal derivatives and the other class is associated with both the depth-dependent and the azimuth-dependent modes and their range and azimuth derivatives. The coupled integral equations for the scattering amplitudes are solved using a method previously developed for a 2-D integral equation coupled mode approach. The method is applied to several 3-D environments including single and multiple seamounts. The effect of neglecting various types of coupling integrals is examined. [Work supported by ONR Code 322 OA.]

10:00

2aUWa3. Massively parallel structural acoustics for forward and inverse problems. Timothy F. Walsh (Computational Solid Mech. and Structural Dynam., Sandia National Labs., PO Box 5800, MS 0380, Albuquerque, NM 87185, tfwalsh@sandia.gov) and Wilkins Aquino (Civil and Environ. Eng., Duke Univ., Durham, NC)

Three-dimensional structural acoustic simulations on highly complex structural models that are immersed in infinite and semi-infinite acoustic domains typically lead to large numbers of degrees of freedom that cannot be solved with commercial software packages. Typical applications include underwater acoustic simulation of submerged structures, and reverberation testing of aerospace structures. Many of these applications of interest involve large acoustic domains and complex 3D structures, thus making a finite element solution an attractive option. In addition, unknown parameters in the models can be mitigated with the solution of inverse problems. In this talk, we will discuss recent research efforts in Sierra-SD in the area of structural acoustics and will also present a partial differential equation (PDE) constrained optimization approach for solving inverse problems in structural acoustics that uses Sierra-SD for solving the forward and adjoint problems. Inverse problems are commonly encountered in structural acoustics, where accelerometer and/or microphone pressures are measured experimentally, and it is desired to characterize the acoustic sources, material parameters, and/or boundary conditions that produced the measurements. With a PDE-constrained optimization approach, the scalability of Sierra-SD can be leveraged for solving inverse problems. We will present results on the application of Sierra-SD on several large-scale structural acoustic applications examples of interest.

10:20

2aUWa4. Three-dimensional modeling in global acoustic propagation. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

The ocean is nearly transparent for acoustic propagation at low frequencies (<100 Hz), leading to the detection of signals (seismic events, volcanoes, and man-made signals) at distances as large as the ocean basin. Historically, basin acoustic modeling has neglected out-of-plane effects and has been performed with the model computed in the range/depth plane for multiple radials following geodesics (Nx2D). Both oceanographic and bathymetric features can lead to out-of-plane effects. In this paper, a summary of computational approaches to this problem will be presented, including vertical-mode, horizontal ray hybrid approaches, and full-3D Parabolic Equation modeling. Out-of-plane effects include refraction and diffraction—which have different effects as well as different approaches to modeling. Experiments where 3D propagation effects were significant will be presented within this context, including Perth-Bermuda (1960), the Heard Island Feasibility Test (1993), and a recent seismic tomography test off the coast of Japan (2015). Three physics mechanisms will be addressed: horizontal deflection due to mesoscale eddies and fronts, reflection from islands (refraction), and diffraction behind bathymetric edges.

10:40

2aUWa5. Broadband acoustic wave propagation in three-dimensional shallow waveguide with variable sound speed profile and boundary roughness. Mohsen Badiy (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiy@udel.edu)

Propagation of broadband acoustic signals in shallow water environment is a complex four-dimensional problem that needs to be addressed with input from the spatial and temporal physical parameters of waveguides with rough boundaries. To construct a numerical model of this four-dimensional problem, methods such as the Parabolic Equation (PE), Horizontal Rays and Vertical Modes, 3D Ray Method, and Nx2D PE have been utilized in recent years. However, data/model comparison still remains a challenge and accurate comparison between measured and modeled acoustic fields in the waveguide is badly needed. Lack of environmental input to use for modeling is one reason, but with proper sampling of environment it may be overcome by novel experimental design based on the knowledge of waveguide physics. Acoustic frequency can also be utilized as one of the key parameters to simplify the problem and adopt strategies in conducting calibrated experiments. In this paper, we provide a broad view of recent advancements in three-dimensional acoustic wave propagation in shallow water waveguides in the presence of variable volumetric and boundary conditions. The effect of broadband acoustic wave center frequency and bandwidth with respect to the scale of environmental variability is also discussed. [Work supported by ONR 322 OA.]

Contributed Papers

11:00

2aUWa6. Propagation over a rigid sea ridge using a three-dimensional finite element model. Fiona Cheung and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, PO Box 8029, Austin, TX 78713, fiona.cheung@utexas.edu)

A three-dimensional finite element model is developed to describe propagation over a rigid underwater sea ridge. Finite element models are attractive benchmark solutions since they include all orders of scattering as well as refraction. In this case, the model is calculated using an out-of-plane wavenumber decomposition technique which utilizes a series of two-dimensional models to calculate a fully three-dimensional model in longitudinally invariant environments. [Isakson et al., J. Acoust. Soc. Am. Express Letters, 136:EL206-211, 2014.] The time-harmonic model is then extended to the time domain via Fourier synthesis in order to more fully understand the dynamics of modal refraction and propagation over the ridge. The model is

compared with coupled mode solutions from the University of Rhode Island. [Work supported by ONR, Ocean Acoustics.]

11:20

2aUWa7. Time-domain reverberation modeling for rough bottom consisting of polygon facets. Youngmin Choo (Defense System Eng., Sejong Univ., Seoul National University, 1, Gwanak-ro, Gwanak-gu, Seoul, Seoul 151 - 744, South Korea, sks655@snu.ac.kr) and Woojae Seong (Seoul National Univ., Seoul, South Korea)

In reverberation modeling for rough bottom, a surface integration is conducted along elemental scattering areas with repetitive uses of propagation and scattering strength models and a summation of scattered signals from the scattering areas provides a synthetic reverberation signal in time domain. In particular, when roughness is on a flat or sloping bottom, numerical integration schemes including quadrature by parts can be used with elemental scattering areas, which are small enough to obtain a converged reverberation

signal. However, this standard approach is unavailable for a bottom having irregular geometry since the bottom cannot be divided into small element scattering areas. To acquire a stable reverberation signal by the irregular bottom, we derive an analytic integration of scattered signal for polygon facet by using Stokes' theorem while approximating the bottom with combination of polygon facets. In this approach, a delay difference in an elemental scattering area is considered whereas a representative delay is used for the elemental scattering area in the standard approach. Results from two different reverberation models are compared and the scheme using analytic integration shows a converged reverberation signal even with large elemental scattering areas.

11:40

2aUW8. Computation of sound field, reflected from ideal non-flat boundary, and reflected and refracted from non-flat boundary of two different media in parabolic approximation. Nikolai Maltsev (Lookomorie, 1467 Leafree cir, San Jose, CA 95131, nick_e_maltsev@yahoo.com)

Four Euler equations for harmonic sound field $\rho c^2(\nabla, v) = i\omega p, \nabla p = i\omega p v$ can be reduced to two equations for sound pressure p and radial velocity v_r in cylindrical coordinate system (r, φ, z) : $\partial p / \partial r = i\omega p v_r$, $\partial v_r / \partial r = i/(\omega \rho)(k^2 + (1/r^2)\partial^2/\partial \varphi^2 + \partial^2/\partial z^2)p - 1/r v_r$. Solutions for this system of equations in a set of 3-D local normal modes is constructed. A stable numerical method was created, for computation of sound field,

reflected from ideal non-flat boundary, and reflected and refracted from non-flat boundary of two different medias, in parabolic approximation. Solutions of different model problems are presented.

12:00

2aUW9. Three dimensional acoustic parabolic equation based on Chisholm approximation with the splitting denominator. Keunhwa Lee (Defense Systems Eng., Sejong Univ., Neungdong-ro 209, Seoul 05006, South Korea, nasalkh2@sejong.ac.kr) and Woojae Seong (Ocean Eng., Seoul National Univ., Seoul, South Korea)

We propose the generalized form of three dimensional acoustic parabolic equation (3DPE) based on Chisholm approximation [Chisholm, Math. Comp. 27, 841-848 (1973)] of a rational approximant for two variables, and the splitting denominator assumption. The proposed form has wider angle accuracy to the inclination angle of ± 620 from the range axis of 3DPE at the bearing angle of 450. Moreover, the splitting denominator makes the split-step algorithm with finite-difference depth solver more efficient in that the 3DPE can be easily transformed into the tridiagonal matrices system. One drawback of this method is the increase of the phase error in the evanescent region, but should be practically remedied by several skills. In this study, the comparative study of other PE approximations will be conducted based on the phase error analysis. Also, numerical examples with three dimensional problems will be given for the performance and benchmark test.

2a MON. AM

MONDAY MORNING, 26 JUNE 2017

ROOM 309, 9:20 A.M. TO 11:40 A.M.

Session 2aUWb

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 III

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

9:20

2aUWb1. Dynamic imaging of a gravity wave caused by laser-induced breakdown in a fluid waveguide using acoustic waves. Tobias van Baarsel, Philippe Roux (Université Grenoble-Alpes, Université Grenoble Alpes ISTerre, Grenoble 38000, France, tobias.van-baarsel@univ-grenoble-alpes.fr), Barbara Nicolas (Créatis, Villeurbanne Cedex, France), Jérôme Mars (Université Grenoble-Alpes, Grenoble, France), Julien Bonnel, and Michel Arrigoni (ENSTA, Brest cedex 9, France)

The dynamic imaging of a gravity wave propagating at the air-water interface is a complex task that requires the sampling of every point at this interface. Using two source-receiver vertical arrays facing each other in a shallow water environment, we manage to isolate and identify each multi-reverberated eigenray that interacts with the air-water interface. The travel-time and amplitude variations of each eigenray are then measured during the crossing of the gravity wave. In this work, we present an ultrasonic experiment in a 1 m-long, 5 cm-deep waveguide. At frequencies in the MHz range, the waveguide transfer matrix is recorded 100 times per second between two source-receiver arrays while a low-amplitude gravity wave is generated by a laser-induced breakdown above the water surface. The breakdown causes a blast wave that interacts with the air-water interface and penetrates into the water, creating ripples at the surface. This event is easily controlled and therefore repeatable. The inversion performed from a few thousand eigenrays leads to accurate imaging of the dynamic of the air-water interface, using either the travel-time or the amplitude variation.

2aUWb2. Statistical inference for source localization using multi-frequency machine learning. Haiqiang Niu (Marine Physical Laboratory, Scripps Inst. of Oceanogr., San Diego, CA), Peter Gerstoft, and Emma Reeves (Marine Physical Laboratory, Scripps Inst. of Oceanogr., Univ. of California, La Jolla, CA, pgerstoft@ucsd.edu6)

As a classification problem in machine learning, source localization is solved by training a feed-forward neural network (FNN) on ocean acoustic data. The FNN is fed with normalized sample covariance matrices (SCMs). The output of network, which represents the probability for range, is used to determine the source ranges. Since it is a data-driven method, no acoustic propagation models are needed. As shipping noise has a broad frequency band, an approach of statistical inference for source localization is presented by taking advantage of multi-frequency information. It is demonstrated by the vertical array data from Noise09 experiment.

Contributed Papers

10:00

2aUWb3. Effect of dispersion on the convergence rate for Green's function retrieval. John Y. Yoritomo and Richard Weaver (Phys., Univ. of Illinois at Urbana-Champaign, 1110 West Green St., Urbana, IL 61801-3080, yoritom2@illinois.edu)

Much information about wave propagation in a variety of structures has been obtained from Green's function retrieval by noise correlation. Here we examine how dispersion affects Green's function retrieval and, in particular, its signal-to-noise ratio (SNR). On recalling how the inherent spread of a signal due to band limitation is augmented by spread due to dispersion and propagation distance, and how both affect amplitude, we argue that SNR in highly dispersive media can be substantially lowered by strong dispersion. We argue this is most relevant for gravity waves over large propagation distances in the ocean or atmosphere. In particular, we argue that dispersion could explain recent retrieval failure from surface gravity wave noise in the ocean. Lastly, we consider methods to ameliorate the poor SNR due to dispersion. We use numerical simulation to substantiate our analytic results.

10:20–10:40 Break

10:40

2aUWb4. Estimating the speed of poroelastic interface waves using ambient noise. David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., PO Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca), Len Zedel (Phys. and Physical Oceanogr., Memorial Univ. of NF, St. John's, NF, Canada), and Alex E. Hay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Pairs of hydrophones were buried at mid-tide height in a 1:10 sloped mixed gravel and coarse sand beach and used to make ambient noise recordings over a period of three weeks in Advocate Harbour, Nova Scotia, in the Bay of Fundy. The pairs were arranged in vertical and horizontal configurations and recorded pressure time series, power spectral density, and vertical and horizontal coherence measurements of the noise field in the seabed. A nearby suite of oceanographic instruments measured the water level, ocean wave properties, bed dynamics and weather during the experiment. The measured noise between 1 Hz and 1 kHz was dominated by poroelastic interface waves generated by plunging surf on the beach face. The speed of the compressional component of the interface wave was estimated by cross-correlating the noise recorded on the across-shore oriented pair of sensors while the unconsolidated sediment was water-saturated as well as drained. Additionally, the increasing and decreasing overburden pressure due to the rising and falling 10-m tide was found to drive a respective increase and

decrease in the poroelastic interface wave speed. A buried acoustic source was used to directly measure the compressional wave speed in the seabed on the across-shore array.

11:00

2aUWb5. Passive bottom reflection-loss estimation using ship noise and a vertical line array. Lanfranco Muzi and Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, muzi@pdx.edu)

An existing technique for passive bottom-loss estimation from natural marine surface noise (generated by waves and wind) is adapted to use ship generated noise. The original approach (based on beamforming of the noise field recorded by a vertical line array of hydrophones) is retained. However, the field generated by a passing ship must be processed preliminarily, in order for it to show features that are similar to those of the natural surface-noise field and therefore become amenable to the technique. A necessary requisite is that the ship position, relative to the array, vary over as wide a range of steering angles as possible, ideally passing directly over the array to ensure coverage of the steepest angles. The methodology is illustrated through simulation and applied to experimental data.

11:20

2aUWb6. Channel impulse response arrival uncertainty using source of opportunity for tomography. Kay L. Gemba, Jit Sarkar, Jeffery D. Tippmann, William S. Hodgkiss, Bruce Cornuelle, William A. Kuperman (MPL/SIO, UCSD, Univ. of California, San Diego, 8820 Shellback Way, Spiess Hall, Rm. 446, La Jolla, CA 92037, gemba@ucsd.edu), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Passive acoustic tomography exploits the acoustic energy generated by sources with unknown spectral content such as sources of opportunity (e.g., cargo ships) to study the ocean. The recording at each sensor within a vertical line array (VLA) is the channel impulse response (CIR) convolved with the noise generated by the moving random radiator. Using an incremental approach, we estimate the source signal locally at three VLAs by beamforming on the direct ray-path to deconvolve each CIR. CIR arrival uncertainty is inversely proportional to the bandwidth of the source and SNR, the latter estimated from the deconvolved time domain waveform. Over the 10 min source track, we present the time evolution of CIR arrival uncertainty computed at three VLAs horizontally separated by 1.5 km and discuss constraints on integration time. Data are presented using the Noise Correlation 2009 Experiment and application to the Santa Barbara Channel Experiment 2016 is discussed.

Meeting of the Standards Committee Plenary Group

to be held jointly with the meetings of the

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

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

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

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

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

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

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

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

D. A. Preves and C. Walber, U.S. Technical Co-advisors for IEC/TC 29, Electroacoustics
*Starkey Hearing Technologies, 6600 Washington Ave., S., Eden Prairie, MN 55344 (D. Preves)
PCB Piezotronics, Inc., 3425 Walden Avenue, Depew, NY 14043 2495 (C. Walber)*

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

The meeting of the Standards Committee Plenary Group will follow the meeting of Accredited Standards Committee S2, which will be held on Sunday, 25 June 2017 from 5:00 p.m. to 6:00 p.m.

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

Monday, 26 June 2017	11:00 a.m. – 12:15 p.m.	S12, Noise
Monday, 26 June 2017	2:00 p.m. – 3:00 p.m.	ASC S3/SC 1, Animal Bioacoustics
Monday, 26 June 2017	3:15 p.m. – 4:30 p.m.	ASC S3, Bioacoustics
Monday, 26 June 2017	4:45 p.m. – 5:45 p.m.	ASC S1, Acoustics

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

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

<u>U.S. TAG Chair/Vice Chair</u>	<u>TC or SC</u>	<u>U.S. Parallel Committee</u>
ISO		
R. D. Hellweg, Jr., Chair P. D. Schomer, Vice Chair	ISO/TC 43 Acoustics	ASC S1 and ASC S3
R. D. Hellweg, Jr., Chair P. D. Schomer, Vice Chair	ISO/TC 43/SCI Noise	ASC S12
M. A. Bahtiarian, Chair W. Madigosky, Chair M. L'vov, Chair	ISO/TC 43/SC 3 , Underwater acoustics ISO/TC 108 Mechanical vibration, shock, and condition monitoring ISO/TC 108/SC2 Measurement and evaluation of mechanical vibration, and shock as applied to machines, vehicles and structures	ASC S1, ASC S3/SC 1, and ASCS12 ASC S2 ASC S2
D. J. Evans, Chair	ISO/TC 108/SC3 Use and calibration of vibration and shock measuring instruments	ASC S2
D. D. Reynolds, Chair	ISO/TC 108/SC4 Human exposure to mechanical vibration and shock	ASC S2 and ASC S3
D. J. Vendittis, Chair	ISO/TC 108/SC5 Condition monitoring and diagnostics of machine systems	ASC S2
IEC		
D. A. Preves and C. Walber, U.S. Technical Co-advisors	IEC/TC 29 Electroacoustics	ASC S1 and ASC S3

Meeting of Accredited Standards Committee (ASC) S12 Noise

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

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

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

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1, Noise, and ISO/TC 43/SC 3, Underwater acoustics, take note—that the meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Monday, 26 June 2017.

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

2a MON. AM

Exhibit

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.

2pAAa**Architectural Acoustics: New Measurement and Prediction Techniques at Low Frequencies in Buildings**

James E. Phillips, Cochair

Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Bert Roozen, Cochair

Physics and Astronomy, KU Leuven, Leuven, Belgium

Herbert Muellner, Cochair

*Acoustics and Building Physics, Federal Institute of Technology TGM Vienna, Wexstrasse 19-23, Vienna A-1200, Austria***Chair's Introduction—1:15*****Invited Papers*****1:20****2pAAa1. An introduction to the new edition of AISC Design Guide 11 “Vibrations of Steel Framed Structural Systems due to Human Activity.”** Eric E. Ungar (Acentech, 33 Moulton St., Cambridge, MA 02138-1118, eungar@acentech.com)

The 2016 edition of Steel Design Guide 11, like the first (1997) edition, “Floor Vibrations due to Human Activity,” presents relatively easily used means for predicting the vibrations of rectangular bays of floors of steel-framed construction due to typical walking and for assessing the acceptability of these vibrations to personnel. However, the new edition includes better representations of the forces associated with footfalls and improved validated methods for prediction of the structural responses. In addition to floors of buildings, it also addresses footbridges and stairs. It distinguishes between “low frequency” and “high frequency” structures; the former tend to vibrate nearly steadily at resonance due to typical walking, whereas the latter tend in essence to respond to a series of separate footfall impulses. The new edition presents not only expressions for the expected peak velocities and accelerations, but also includes relations for evaluating the acceptability of vibrations for equipment and activities whose criteria are expressed in terms of one-third-octave-band or narrow-band values. It also notes how the different response measures are affected differently by changes in the structural mass and stiffness. An extensive chapter provides advice on the application of finite-element analysis.

1:40**2pAAa2. An alternative statistical method for characterizing low-frequency environments in sensitive laboratory settings.** Byron Davis (Vibrasure, 1015 Florida St., San Francisco, CA 94110, byron@vibrasure.com)

Many practitioners are familiar with the collection and expression of complex noise data as Ln statistical spectra. Given suitable instrumentation, it is easy to develop Ln statistics for low-frequency sound and vibration environments, as well. However, the traditional (instrument-generated) Ln datasets have some shortcomings when it comes to understanding some environments, especially in highly sensitive settings like research laboratories. The emergence of mass data storage and manipulation tools has provided an opportunity to employ a somewhat subtler method to characterize these environments. In this presentation, we will describe this analytical technique, which provides statistical representations similar to the Ln system. However, the technique further allows an explicit invocation of time-scale, allowing the user to choose an arbitrary analytical timescale relevant to sensitive operations or expected environmental transients or other forcings. We will also show examples that demonstrate why the approach might find particular use in technical settings like nanotechnology laboratories and other environments supporting research in the physical sciences.

2:00**2pAAa3. Micro-vibration design for a cutting edge cancer research laboratory at a high-vibration site.** Ahmad Bayat, Thomas Kaytt, and Alana DeLoach (Vibro-Acoust. Consultants, 490 Post St., Ste. 1427, San Francisco, CA 94102, tom@va-consult.com)

State-of-the art medical & material science imaging technologies are pushing the limits of ground vibration tolerance. Modern technical instruments such as Transmission Electron Microscopes are calling for tighter functional vibration limits far below the threshold of human sensitivity and at lower and lower frequencies to facilitate analysis of details on the scale of individual atoms. In one particularly demanding case, an instrument proposed for a university research building called for vibration levels not to exceed 50 μ -in/s at frequencies as low as 1 Hz, and this instrument would be needed while heavy construction is ongoing at nearby surrounding properties. Meeting this strict criterion without limiting the use of the surrounding land required special low-noise instrumentation, an in depth geotechnical analysis of the site, and a comprehensive isolation scheme to achieve up to a 95% reduction in ground vibration. This paper will discuss the steps taken to evaluate the site vibration conditions and the structural isolation & active isolation mounting measures enacted to meet this stringent instrumentation requirement.

2:20

2pAAa4. Numerical analysis on applicability of measurement method according to ISO 16283 in small rooms at low frequencies. Stefan Schoenwald and Armin Zemp (Lab. for Acoustics/Noise Control, Empa Swiss Federal Labs. for Mater. Sci. and Technol., Überlandstrasse 129, Dübendorf 8606, Switzerland, stefan.schoenwald@empa.ch)

The robustness of the new measurement method for sound pressure level in small rooms with less than 25 m³ room volume at low frequencies from 50 Hz to 80 Hz that was recently introduced in the ISO 16283 series on sound insulation measurements in buildings was investigated in an experimental study. This restricted study revealed some potential problems with the method that were already presented, but unfortunately, a further experimental investigation was not possible because of the time- and labor-intensiveness of the conducted experiments. Now the sound level distribution in the room was predicted with a simple analytical modal model and an excellent agreement with the available experimental results was found. With the prediction model, it was possible to refine the results of the experimental study, to extend it to other room geometries, and to revisit and to analyze the identified potential problems on a much more detailed and broader database. In the conference paper, the experimental study and its outcomes are briefly recapitulated, the prediction model and its validation are presented, and new conclusions are drawn based on the findings of the original experimental and the extended numerical study.

2:40

2pAAa5. Sound radiation efficiency of lightweight building constructions—Study on the influence of panel fastening by numerical calculations and laser scanning vibrometry measurements. Maximilian Neusser and Thomas Bednar (Res. Ctr. for Bldg. Phys. and Sound Protection, Technische Universität Wien, Karlsplatz 13/206/2, Vienna 1040, Austria, maximilian.neusser@tuwien.ac.at)

The aim of the presented work is to develop a calculation model for predicting vibration characteristics and consequent measures of the sound radiation efficiency of lightweight building structures including the influences of their connection means. An application of the currently normative covered processes for these structures is excluded in the relevant body of standards. Through the analysis of the velocity distribution on the surface by laser vibrometry and the simultaneously measurement of the introduced vibration energy, transfer functions could be identified. Within the scope of this work, different influencing parameters for the formation of the connecting joint between wall components are identified and their effects on the vibration characteristics are quantified. The measurement results offered not only the identification of parameters but can also be used in the development and validation of the simulation model based on the finite element method. A good correspondence between measurements and results of the introduced numerical model could be achieved. The presented simulation model offers the possibility of the consideration of the identified parameters in the formation of the connecting bodies such as the dimension of the screws, the distance between screws, the tightening torque, and the position of the screws on supporting structures.

3:00–3:20 Break

3:20

2pAAa6. Measuring the sound insulation of an external thermal insulation composite system (ETICS) by means of vibrometry. Daniel Urbán (A&Z Acoust. s.r.o., S.H.Vajanského 43, Nové Zámky 94079, Slovakia, ing.daniel.urban@gmail.com), Bert Roozen (Dep. of Phys. and Astronomy, Soft matter and Biophys., Lab. of Acoust., KU Leuven, Leuven, Belgium), Alexander Niemczanowski (Versuchsanstalt TGM, Fachbereich Akustik und Bauphysik, Wien, Austria), Herbert Muellner (Versuchsanstalt TGM, Fachbereich Akustik und Bauphysik, Vienna, Austria), and Peter Zaľko (A&Z Acoust. s.r.o., Bratislava, Slovakia)

The impact of External Thermal Insulation Composite System (ETICS) on acoustic properties of external walls has been already examined. Probably the most fundamental research was carried out in Germany (Weber). The effect of thickness and dynamic stiffness of ETICS, as well as mass of external plaster, on decreasing of wall sound insulation was demonstrated. In this paper, the mass spring mass resonances (m-s-m) were investigated, in case of a massive external wall with ETICS. Application of ETICS increases the sound insulation of walls in the mass-law dominated frequency range by about 12 dB/oct. However, in the low frequencies ETICS decrease sound insulation due to resonant effects, which become very prominent in traffic noise situations. This contribution presents how vibrometry measurements can be useful for ETICS sound insulation properties measurement.

3:40

2pAAa7. Laser Doppler vibrometry measurement of the radiated sound power of a funicular floor system. Tomas Mendez Echenagucia (Inst. of Technol. in Architecture, Block Res. Group, ETH Zurich, Stefano-Francini-Platz 1, HIB E 46, Zurich, Zurich 8093, Switzerland, mendez@arch.ethz.ch), Bert Roozen (Dept. of Phys. and Astronomy, KU Leuven, Leuven, Belgium), and Philippe Block (Inst. of Technol. in Architecture, Block Res. Group, ETH Zurich, Zurich, Switzerland)

Floor slabs represent a high percentage of the embedded energy in buildings. Lightweight floor systems, such as funicular shell structures, represent an important approach to the reduction of embedded energy in building by reducing material use in a significant way. As the amount of material is reduced, the sound insulation capabilities need to be studied in depth, particularly in the lower frequency range. The high stiffness of the shell structures has been shown with numerical experiments to have great potential for sound insulation in low frequencies. This paper presents laboratory measurements of the radiated sound power an unreinforced concrete funicular floor system in the low frequencies by means of laser Doppler vibrometry (LDV). The presented experiments use the velocities captured by the LDV system, coupled with the Rayleigh integral method to estimate the radiated sound power in an accurate way, without the known mode density problems present in low frequency microphone based measurements. A flat concrete slab of the same mass and dimensions is also tested for comparison. The paper presents the results of the two measurements and outlines guidelines for the acoustic optimization of the funicular floor system.

2p MON. PM

4:00

2pAAa8. An active control method for the spectral homogenization of airborne sound insulation laboratories at low frequencies. Andrea Prato, Alessio Atzori, and Alessandro Schiavi (INRIM, Strada Delle Cacce 91, Torino 10135, Italy, a.prato@inrim.it)

Low frequency sound insulation measurements are affected by large uncertainties and inaccuracies due to the low modal density of small laboratory rooms. For this reason, an automated measuring system for the active spectral homogenization of enclosed spaces at low frequencies has been developed. The aim is to achieve a quasi-diffuse field, reducing the amplitude of room modes, in order to apply standard procedures for sound insulation measurements at low frequencies. The homogenization method is based on the active control of the interference spatial patterns of a system of two loudspeakers using a phase steering technique. The room response spectrum, as function of frequency-dependent phase difference between source signals, is measured in different positions in order to achieve the optimal spectral homogenization. Such technique allows to decrease the modal sound pressure level fluctuations in the source and receiving volumes in the frequency range between 30 Hz and 120 Hz. Based on this, sound insulation measurements at low frequencies on a high performance triple glazing and steel structure are performed and compared with standard ones.

4:20

2pAAa9. Blocked pressure based Transfer Path Analysis (TPA) method to diagnose airborne sound transfer through building partitions. Nikhilesh Patil, Andy Moorhouse, and Andy S. Elliott (Acoust. Res. Ctr., Univ. of Salford, Acoust. Res. Ctr., Newton Bldg., Salford M5 4WT, United Kingdom, n.patil@edu.salford.ac.uk)

Airborne sound transmission through building elements or the sound insulation of the building element is usually rated by its Sound Reduction Index (SRI) or the Sound Transmission Class (STC). SRI/STC quantifies the overall sound transfer but gives no information about how the transfer takes

place and what are the contributions of different sound transfer paths involved. Such problems are fairly common to the vehicle acoustics industry and are generally tackled by TPA techniques. The paper formulates an in-situ Airborne TPA technique to quantify the contributions of different sound transfer paths to the transmitted pressure. The airborne source is characterized by its blocked pressure and its direct measurement is discussed. Results are presented for dual leaf partitions excited by an airborne source. The method has shown to be significantly faster than the Blocked force based TPA method which relies on inverse measurement methods. The accuracy of the method is closely related to the wavelength of incident airborne waves.

4:40

2pAAa10. On the use of finite-element methods to minimize uncertainties in airborne sound insulation measurements in the low frequency range. Francesco Martellotta, Ubaldo Ayr (DICAR, Politecnico di Bari, Via Orabona 4, Bari, Bari 70125, Italy, francesco.martellotta@poliba.it), and Gianluca Rospi (Dipartimento delle Culture Europee e del Mediterraneo, Università della Basilicata, Matera, Italy)

Measuring airborne sound insulation at low frequencies (below 100 Hz) may be very challenging. In fact, to cope with this problem ISO 16283-1:2014 also included a dedicated procedure for smaller rooms, having a volume below 25 m³. However, even in significantly larger rooms, large spatial variations of sound pressure levels may appear, resulting in a measure which is largely affected by the choice of the measurement positions. Considering that simple rooms may be easily modeled using finite-elements tools, it may be advantageous to carry out a preliminary numerical analysis of the spatial distribution of the one-third-octave levels, so to minimize measurement uncertainty. The room space is first subdivided by means of a 3D grid where sound pressure levels are determined using a finite element model. Then, the receiver positions compatible with standard requirements are identified, and, finally, a statistical analysis of the measurement uncertainties is carried out. Comparisons between simulations and measurements are finally illustrated.

2pAAb

Architectural Acoustics: Topics in Architectural Acoustics Related to Application

Kenneth W. Good, Cochair

Armstrong, 2500 Columbia Ave., Lancaster, PA 17601

David C. Swanson, Cochair

Penn State ARL, 222E ARL Bldg., PO Box 30, State College, PA 16804

Contributed Papers

1:20

2pAAb1. The acoustics of rooms for music rehearsal and performance—The Norwegian approach. Jon G. Olsen (Akershus, Norwegian Council for Music Organizations, Akershus musikkråd, Trondheimsveien 50E, Kjeller 2007, Norway, jon.olsen@musikk.no) and Jens Holger Rindel (Oslo, Multiconsult AS, Oslo, Norway)

Each week, local music groups in Norway use more than 10,000 rooms for rehearsal and concert; many of the rooms are in schools. The size of the rooms vary from under 100 m³ to over 10,000 m³. The users cover a broad variety of music ensembles, mostly wind bands, choirs, and other amateur ensembles. Since 2009, the Norwegian Council for Music Organizations («Norsk musikkråd») has completed more than 500 room acoustical measurement reports on rooms used for rehearsal and concert. The measurements include reverberation time, the strength parameter G, and background noise. All the reports are made available online in a Google Map. The analysis shows that 85% of the rooms do not comply with the Norwegian Standard NS 8178:2014 and are evaluated more or less unsuitable for the purpose for acoustical reasons. The important criteria are volume, room dimensions, reverberation, acoustic treatment of surfaces, and background noise. In particular, the importance of volume is clearly documented. Analysis of room strength indicates that this also is an essential factor for this type of rooms. The systematic collection of acoustic reports gives important background for recommendations on how to build or refurbish rooms for music in schools and cultural buildings.

1:40

2pAAb2. Perception of acoustic comfort in large halls covered by transparent structural skins. Monika Rychtarikova (Faculty of Architecture, KU Leuven, Hoogstraat 51, Gent 9000, Belgium, Monika.Rychtarikova@kuleuven.be), Daniel Urban (A&Z Acoust., Bratislava, Slovakia), Magdalena Kassakova (Faculty of Civil Eng., Dept. of Bldg. Construction, STU Bratislava, Bratislava, Slovakia), Carl Maywald (Vector Foiltec, Bremen, Germany), and Christ Glorieux (Phys. and Astronomy, Lab. of Acoust., KU Leuven, Leuven, Belgium)

Large halls, such as shopping malls, atria, or big entrance halls, often suffer from various acoustic discomfort issues, which are not necessarily caused by extremely high noise levels. Due to the large size of halls and consequently the long trajectories that sound waves travel between the source, interior surfaces, and the receiver, sound reflections arriving from surrounding surfaces are not as strong as they would be in smaller rooms. Reports in literature and comments by users of large halls concerning acoustic discomfort in large halls refer mainly to continuous reverberation related noise. Therefore, quantification of the acoustic comfort by the reverberation time, which is related to the average absorption of interior surfaces and by the equivalent sound pressure level, which in a large space is dominated by direct sound, is not adequate to describe the global acoustic comfort or soundscape. Based on statistical noise analysis on auralized soundscapes, this article proposes a set of measurable monaural and binaural acoustic

parameters that adequately describes the acoustic comfort in large halls. The study is focusing on rooms covered by traditional materials, such as glass, plexiglass, etc., and ETFE (ethylene tetrafluoroethylene) foil structures.

2:00

2pAAb3. Acoustical design of diffusers in concert halls using scale models. HyunIn Jo, Hyung Suk Jang, and Jin Yong Jeon (Dept. of Architectural Eng., Hanyang Univ., Hanyang University, Seoul, Seongdong-gu 133-791, South Korea, best2012@naver.com)

A design process of diffusing surfaces has been verified by comparing with acoustic parameters in different scale models of a concert hall. A 1:50 scale model was used to determine locations of the diffusers as the main diffusing surfaces, and a 1:25 scale model was used to determine the structural height and density of the stage diffuser by examining the amount of diffusion on the stage. In addition, a 1:10 scale model was used to design the exact shapes of the diffusers by measuring the scattering and diffusion coefficients of them. The diffusers were installed on the concert hall stage and sidewalls of the auditorium; acoustic parameters, such as EDT, G, C80, and Np were measured to examine the amount of diffusion in the hall. As a result, the relative standard deviations of the parameters RT, EDT, and G decreased, whereas Np values increased. In the case of Np, the change was consistent with the variation of the diffusion amount. In conclusion, when designing a diffusing surface in a concert hall, the scale-model measurement is essential to determine the location and amount of the effective diffuser, as well as the direction of the diffuser geometry.

2:20

2pAAb4. The acoustic design of the new University of Iowa Voxman School of Music. Russell A. Cooper and Steven Schlaseman (Jaffe Holden Acoust., Inc., 114A Washington St., Norwalk, CT 06854, rcooper@jaffeholden.com)

The University of Iowa Voxman School of Music opened for students in September 2016. Necessitated by a flood of the Iowa river in 2008 that condemned the original building, this ground up new building in the center of Iowa City features a 700 seat concert hall, a 200 seat recital hall, a 75 seat organ recital hall; band, orchestra, chamber music, and choral rehearsal rooms; an opera studio; recording, percussion, and electronic music suites; teaching studios; practice rooms; library; classrooms and social spaces. Designed on a single city block and stacked 6-1/2 floors high, the design presented many sound isolation challenges. The varied musical pedagogy also required that each performance and rehearsal space have adjustable acoustics. The concert hall, the most public space in the facility houses a brand new 3,883 pipe Klais organ. The TheatroAcoustic ceiling in the hall is a beautiful example of coordination of all design disciplines: acoustics, rigging, lighting, sound, HVAC, fire suppression, recording, and aesthetics. This paper presents the criteria and design of the facility, the innovations, as well as tried and true and the results of acoustic measurements.

2:40

2pAAb5. Effect of orchestra absorption on the reverberation time in concert halls. Sung Min Kim, Hyung Suk Jang, and Jin Yong Jeon (Dept. of Architectural Eng., Hanyang Univ., Sung dong-gu Wangsimni-ro 222, Seoul 133-791, South Korea, rainbear0622@gmail.com)

Based on 1:10 scale models and computer simulations performed in this study, the number of players in an orchestra and their sound absorption are suggested as important factors affecting the sound in the hall. For the computer simulation, scale models of players were constructed and the absorption rates were measured in the reverberation chamber. The simulation results suggest that a sound absorption rate per person is affected by the density of the occupied area of the players, and the sound absorption coefficient is determined for the space per player. When comparing two spaces of different sizes with the same number of players, the reverberation time of the auditorium is largely affected by the players when the space is larger. If more than 5% of the sound absorbing area of the hall is occupied by players, the reverberation time is remarkably reduced because of the sound absorption of the players. Therefore, the results from this study can be utilized to more accurately predict the reverberation and clarity in the acoustic design.

3:00

2pAAb6. Variable room design in office spaces by the use of sound insulating curtains. Jonas Schira (Sales Manager Acoust., Gerriets GmbH, Im Kirchenhürstle 5-7, Umkirch 79224, Germany, jschira@gerriets.com)

The use of open space design in offices has been widely used over the last years. The idea of a collaborative working zone for more than just two employees has both advantages and disadvantages. One of the disadvantages is that private meeting zones or thin tanks get lost when a simple open space architecture is used. This fact can be solved by using variable sound insulating components to create variable zones inside the open space architecture. In the field of theater, techniques such variable sound insulating elements have been used since the last decades in the form of sound insulating curtain systems. To create a good sound insulation, more than just one layer of heavy and highly absorptive fabric is needed. By combining reflective and absorptive materials and using a special track system and ceiling connection, a sound insulation of up to $R_w' = 25$ dB can be achieved. This lecture will illustrate the use of sound insulation curtain systems as a part of an innovative open space office design. Also, the technical aspects of building and installing such a system will be discussed.

3:20–3:40 Break

3:40

2pAAb7. Sound masking in office environments—Trade-off between masking effect and user acceptance. Noemi D. Martin and Andreas Liebl (Acoust., Fraunhofer Inst. for Bldg. Phys. IBP, Nobelstrasse 12, Stuttgart 70596, Germany, noemi.martin@ibp.fraunhofer.de)

Solving cognitive tasks is adversely affected by speech sound (Irrelevant Speech Effect). In particular, the interfering potential of speech sound is high in open-plan office environments. *Sound masking* is one of the most effective methods to reduce the disturbing speech sound by covering certain fractions of it with a masking signal. Since the most effective masking signals are subjectively perceived as annoying themselves, the goal of this work was to develop a masking signal which fulfills both, a good masking effect and a high user acceptance. To achieve this goal, qualitatively different masking signals with varying structural properties were developed. The efficacy of these signals was tested in a laboratory experiment using a cognitive task (serial recall). In addition, a subjective evaluation of the signals

(loudness, annoyance, etc.) was carried out. In the second part of this work, a questionnaire was developed, which can be used to evaluate the acceptance and the subjectively assessed effectiveness of newly developed masking signals by their potential users. Using this questionnaire, the newly developed masking signals were evaluated by persons working in open-plan offices. Findings and implications for practical usage are presented.

4:00

2pAAb8. Using acoustical modeling software to predict speech privacy in open-plan offices. Valerie Smith (Charles M. Salter Assoc., 130 Sutter St., San Francisco, CA 94104, valerie.smith@cmsalter.com)

Speech Privacy Index is one of the commonly used metrics to discuss an occupant's acoustical comfort in an open-plan office. In existing open-plan offices, the Articulation Index can be measured using a qualified sound source; the Privacy Index can then be calculated from the Articulation Index. However, in the case of future office planning, the Privacy Index must be estimated. Using the ODEON acoustical modeling software, we estimated the Privacy Index in the open-plan section of our office using the existing room dimensions, material finishes, and background noise levels. The ODEON estimates were then compared with "real world" measurements of the same space. This paper summarizes the differences between the estimated and measured speech privacy index levels.

4:20

2pAAb9. The acoustic design of a conference room: From sketches to measurements. Fabio Sicurella (Planair SA, Crêt 108, La Sagne 2314, Switzerland, fabio.sicurella@planair.ch), Gino Iannace (Università della Campania "Luigi Vanvitelli", Aversa, Italy), Perla Colamesta (Planair SA, La Sagne, Switzerland), and Matteo Gentilin (Stähelin architectes SA, Delémont, Switzerland)

This paper reports the multidisciplinary approach applied for the design of a new conference room in Switzerland. This conference room belongs to a large new educational building realized in Delémont (CH) in 2016. The architectural approach focused on the choice of the room's shape and materials (mainly timber) while the acoustic treatments had to provide a good insulation from exterior and adjacent rooms as well as a good intelligibility, clarity and speech definition. Simulations run with the software Odeon allowed to foresee the main acoustic indicators (T30, EDT, C80, D50, and STI) and therefore to optimize the dimension and position of the acoustic treatments for different occupancy rate. Moreover, the acoustic treatment enforced the vocal emission of the speaker without amplification systems. A measurement campaign at the end of the building construction confirmed the good acoustic quality of the conference room as well as some discrepancies with the forward analysis due to some changes during construction. The results of a survey are also reported in this paper in order to better understand the actual acoustic feeling of the users.

4:40

2pAAb10. Acoustic design of a new museum. Attila B. Nagy and András Kotschy (Kotschy and Partners Ltd., Almos vezér u. 4., Torokbalint 2045, Hungary, attila.nagy@kotschy.hu)

In this paper, we report on the acoustic design of a new museum. Besides the general exhibition space, the museum building includes three larger multipurpose halls that will give place for lectures and diverse events. Defining the acoustic requirements of the multipurpose hall is always a hard to reach compromise. We demonstrate the results of the international cooperation of acoustic engineers and architects, and show how computer aided modeling helped during all design phases to achieve the fine-tuned final stage.

2pAAc

Architectural Acoustics: Perceptual Effects Related to Music Dynamics in Concert Halls

Tapio Lokki, Cochair

Computer Science, Aalto University, POBox 13000, Aalto 00076, Finland

Michelle C. Vigeant, Cochair

*Graduate Program in Acoustics, The Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802**Invited Papers*

1:20

2pAAc1. How (and why) does every Concert Hall “wake up” differently? Eckhard Kahle, Evan Green, Fabian Knauber, Thomas Wulfrank, and Yann Jurkiewicz (Kahle Acoust., Ave. Molière 188, Brussels 1050, Belgium, kahle@kahle.be)

Masking is a loudness-dependent, non-linear process. Conversely, the process of *unmasking* is significant when considering the perceptual effects of music dynamics: the audibility of reflections relative to the direct sound is loudness-dependent, as shown by Wettschurek in the 1970s. To summarize his description of an orchestral crescendo: in *pianissimo*, all sources are clearly localized and the auditory image is fully frontal; then, as loudness increases, the room increasingly *wakes up* as reflections progressively become unmasked; and, finally, in full *forte*, the room is present, and ideally, the listener should be enveloped by sound. Wettschurek’s description implies a linear increase in spatial impression with loudness; however, the effect of musical dynamics is different in every hall, determined by the unique reflection sequence at a particular listening position. As a consequence, different rooms can lose clarity at different loudness levels, not necessarily only during saturation at *fortissimo*. Recent research by Lokki indicates that the *signature* of a room is determined by the details of the early room response. The links between the signature of a room and the unmasking of reflections will be discussed.

1:40

2pAAc2. Links between spatial impulse response and binaural dynamic responsiveness in concert halls. Robert Essert (Sound Space Vision, 2 Tay House, 23 Enterprise Way, London SW181FZ, United Kingdom, bob.essert@soundspacevision.com)

Recent research at Aalto University suggests that binaural dynamic responsiveness (BDR) in concert halls is (a) desirable and (b) a result of interaction between the emphasis of higher overtones at stronger dynamic levels and the early response of the room; also that tall, narrow, parallel sided “shoebox” halls have a greater degree of BDR than wider non-shoebox geometries. We have been analyzing the time evolution of spatial impulse responses and their connection to room geometry. In a recent paper, we looked at how wall tilt affects the growth of lateral energy in the room. In this paper, we consider how the shape of forward integrated lateral energy in a spatial impulse response may affect perceived strength, spaciousness, and binaural dynamic responsiveness.

2:00

2pAAc3. Towards a common parameter characterizing the dynamical responsiveness of a concert hall. Tapio Lokki (Comput. Sci., Aalto Univ., POBox 15500, Aalto 00076, Finland, Tapio.Lokki@aalto.fi) and Jukka Pätynen (Comput. Sci., Aalto Univ., Espoo, Finland)

Dynamic responsiveness is an important feature in room acoustics for making music more enjoyable. The concert hall should render the most silent pianissimos audible everywhere, including the last row, yet support fully the loudest fortissimos. The realization of these effects is impossible to objectively quantify by analyzing solely the measured impulse responses. Therefore, the analysis should be coupled with additional information related to music and dynamics. These factors can include directivities of both sources and listeners, and the spectral changes in the source signals and hearing sensitivity according to the sound level. With such effects combined to the information obtained from the conventional impulse response, the dynamic responsiveness could be objectively measured. This paper presents a proposed analysis method which is then applied to a variety of measured concert halls. In addition, the results show what magnitude of differences in dynamic responsiveness could be found between concert halls.

2:20

2pAAc4. Effect of concert hall acoustics on tonal consonance of orchestra sound. Jukka Pätynen (Dept. of Comput. Sci., Aalto Univ. School of Sci., Konemiehentie 2, Espoo FI02150, Finland, jukka.patynen@aalto.fi)

The sound of pitched orchestral instruments consists of harmonic frequencies which, in performances, are transmitted over room acoustics. The amplitude relations of the harmonic peaks affect the timbre of one tone. When two or more notes are played together, the effect of consonance and dissonance becomes prominent. The degree of consonance for intervals of musical pitches have been explained by the frequency separation of their harmonic components in relation to the width of critical bands in human hearing. When the playing dynamics is varied, the changes in the instruments’ spectral envelopes are foreseen to alter also the consonance of simultaneous notes.

Furthermore, the room acoustics influence the overall harmonic spectra conveyed to the listeners. This paper presents experiments on the tonal consonance of orchestra instruments at contrasting dynamic levels in various concert halls. By combining binaural hall measurements and anechoic instrument recordings with a consonance-estimating model, the following hypothesis is investigated: do the acoustics of concert halls change the orchestra sound's consonance in different music dynamics?

2:40–3:00 Panel Discussion

MONDAY AFTERNOON, 26 JUNE 2017

ROOM 208, 3:20 P.M. TO 5:20 P.M.

2pAAd

Architectural Acoustics: Room Acoustics Design for Improved Behavior, Comfort, and Performance I

Nicola Prodi, Cochair

Dept. of Engineering, University of Ferrara, via Saragat 1, Ferrara 44122, Italy

Kenneth P. Roy, Cochair

Building Products Technology Lab, Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603

Invited Papers

3:20

2pAAd1. Acoustic comfort for hypermarket cashiers: Problems and solutions. Francesco Martellotta, Sabrina Della Crociata, Antonio Simone, and Michele D'Alba (DICAR, Politecnico di Bari, Via Orabona 4, Bari, Bari 70125, Italy, francesco.martellotta@poliba.it)

An acoustic investigation carried out in a large hypermarket pointed out that several acoustic “zones” could be identified. Among them, the most critical was the checkout area, where cashiers were exposed to many noise sources like customers passing with shopping trolleys, “beeps” due to barcode readers, people voices, packaging and paging messages, and, above all, background music. In fact, as the checkout position usually divides the hypermarket from the shopping arcade, workers in this area are further exposed to music (and noise) from the nearby cafes, restaurants, and stores. As a result background noise L_{A90} is about 66 dB, with peaks (L_{A10}) of about 72 dB. Subjective analysis also reported the highest rate of complaints in this area. Starting from a detailed analysis of the different noise sources and of the room acoustic conditions of the area (including reverberation time and speech transmission index measurements), a set of mitigation actions is analyzed and discussed.

3:40

2pAAd2. Subjective and objective acoustical quality in healthcare office facilities. Murray Hodgson (UBC, SPPH - 2206 East Mall, Vancouver, BC V6T1Z3, Canada, murray.hodgson@ubc.ca)

This paper discusses acoustical quality in 17 healthcare office facilities. A subjective survey assessed office worker perceptions of their environments and satisfaction with the acoustics. Self-reported productivity, well-being, and health outcomes were also captured. Satisfaction was lower with acoustics than with other aspects of IEQ. Satisfaction results were related to room type and the absence or presence of a sound-masking system. Physical acoustical measurements were made in six types of rooms, some with sound-masking systems, to determine the acoustical characteristics, assess their quality and relate them to the building designs. Background-noise levels were measured in the occupied buildings. In the unoccupied buildings, measurements were made of reverberation times, and “speech” levels needed to calculate speech intelligibility indices for speech intelligibility and speech privacy. In open offices, sound-level reductions per distance doubling (DL2) were measured. The results are presented, and are related to room type and partition design. The knowledge gained from this study informs the decision-making of designers and facilities management for upgrades and future design projects.

4:00

2pAAd3. Advances in adjustable acoustics systems in large multi-use performing arts centers in the U.S. Mark Holden, Mathew Nichols, and Carlos Rivera (Jaffe Holden Acoust., 114A Washington St., Norwalk, CT 06896, mholden@jaffeholden.com)

In the fall of 2016, Jaffe Holden opened three large (1800-2500 seats) multi-use halls with excellent acoustic reviews for all performances from symphony performances to highly amplified popular music. New designs of halls in Salt Lake City, Little Rock, and South Texas prove that this uniquely American building type has dispelled the myth that multi-purpose is in fact “no purpose.” Acoustic modeling calculations will be compared and contrasted with completed building measurements of EDT, RT30, C80, BR, and other criteria proving that these halls are excellent quantitatively and qualitatively.

4:20

2pAAd4. Advances in room acoustics design for educational audio studios. John Storyk (MPE, Berklee College of Music, 262 Martin Ave., Highland, NY 12528, john.storyk@wsdg.com)

Improved behavior, comfort, and performance are critical in creating today's audio recording and post production educational facilities. As these facilities continue to grow and become more widespread in high school and secondary learning institutions, the challenges that they present also continue to grow. Fundamental acoustic behavior remains paramount, but unique considerations associated with the teaching aspect of these rooms present some interesting issues. These include isolation issues, internal room acoustic performance, and interfacing with teaching and industry commercial ergonomic needs (among others). This presentation will use the recently completed Berklee College of Music Studio Complex (Boston, MA) as its prime example and should be associated with a technical tour of the studios provided during the conference.

4:40

2pAAd5. Acoustical design for a little big theater: Instituto Brincante. José A. Nepomuceno (Acústica & Sônica, Acustica & Sonica, Rua Fradique Coutinho, 955 cjt 12, São Paulo, São Paulo 05433-000, Brazil, info@acusticaesonica.com.br)

The Brincante Institute located in São Paulo, Brazil, is a space devoted to the study and re-creation of the multitudinous Brazilian artistic manifestations and cultural heritage. The space Brincante occupied from 1990 to 2014 with a theater, shops, and classrooms was demolished to give place for a commercial building. A new building was designed and built

for the "new" Brincante, thanks to the donation of different groups and crowd funding campaign. The new house is a two stories construction that occupies a small land of 200 m². The design challenges were great: tight budget, quality expectations, noisy environment, broad use program, and close neighbor constructions. The range of performances includes acoustic and amplified music, dance to choir. The challenges of acoustic design were the small theater for 100 seats and the rehearsal room, requiring excellent acoustical conditions and high sound isolation due to location specifics. The theater's reduced size helped to keep audience close to players and the sensation of envelopment and clarity. This paper describes the very simple and high performance solutions used for sound isolation and acoustical conditioning. Artists and public received the acoustics with great enthusiasm. The home gives Brincante the opportunity to keep alive.

5:00

2pAAd6. Hospital noise mitigation. Felicia Doggett and Sooch San Souci (Metropolitan Acoust., LLC, 40 W. Evergreen Ave., Ste. 108, Philadelphia, PA 19118, f.doggett@metro-acoustics.com)

A review of the literature today concerning noise in hospitals, both public and peer review, points to a continual increase in noise complaints from medical staff and patients in hospitals. One would expect that things would be getting better by now, but this is not the case. Noise surveys that we have recently conducted suggest that though unsatisfactory conditions are widespread, a list of simple, cost effective solutions has proven to be effective. This paper presents an array of actual projects detailing the development of remedies for noise annoyance in hospitals in an effort to increase the comfort and performance of medical staff and patients.

2p MON. PM

MONDAY AFTERNOON, 26 JUNE 2017

BALLROOM A, 1:20 P.M. TO 4:20 P.M.

2pAAe

**Architectural Acoustics and National Council of Acoustical Consultants:
Student Design Competition**

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

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

The Technical Committee on Architectural Acoustics of the Acoustical Society of America with support from the Robert Newman Student Award Fund, The Wenger Foundation, and the National Council of Acoustical Consultants is sponsoring the 2017 Student Design Competition that will be professionally judged at this meeting.

Design Scenario: A small university has decided to open a multi-purpose facility. It will be located in a densely populated urban setting. It is flanked on both long sides by neighboring buildings. It is to include an auditorium with a balcony, stage house, and orchestra pit. The auditorium will be used as a meeting space and for the school's drama and band programs as well as Broadway productions. The facility will also include a multipurpose rehearsal room which must have easy access to the stage. Music performed and rehearsed in this facility will be chamber ensembles, soloists, jazz ensembles and concert band ensembles.

The submitted designs will be judged by a panel of professional architects and acoustical consultants. An award of USD\$1250 will be made to the submitter(s) of the design judged "first honors." Four awards of USD\$700 each will be made to the submitters of four entries judged "commendation."

2pABa**Animal Bioacoustics, Acoustical Oceanography, Education in Acoustics, and Underwater Acoustics:
Incorporating Underwater Acoustics Research into the Decision Making Process**

Kathleen J. Vigness-Raposa, Cochair

Marine Acoustics, Inc., 2 Corporate Place, Suite 105, Middletown, RI 02842

Michael A. Ainslie, Cochair

*Underwater Tech. Dept., TNO, P.O. Box 96864, The Hague 2509JG, Netherlands***Chair's Introduction—1:15*****Invited Papers*****1:20**

2pABa1. Incorporating basic underwater sound principles into the decision making process. Kathleen J. Vigness-Raposa (Marine Acoust., Inc., 2 Corporate Pl., Ste. 105, Middletown, RI 02842, kathleen.vigness@marineacoustics.com), Gail Scowcroft, Christopher Knowlton, and Holly Morin (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI)

Research on underwater sound is continually advancing. New discoveries on sound in different environments and how sound exposure affects marine animals are just two examples of important ongoing research. To integrate underwater acoustics research into the regulatory process, a fundamental understanding of basic sound principles is required for both producers and regulators. The University of Rhode Island Graduate School of Oceanography teamed with Marine Acoustics, Inc., on the *Discovery of Sound in the Sea* (DOSITS) project to provide scientifically accurate resources on the current knowledge about underwater sound. The project's foundation is a comprehensive website (www.dosits.org). It synthesizes the latest peer-reviewed science on underwater sound in a form that is accessible to a variety of audiences. The site has over 400 pages and is updated semi-annually with newly published information after a thorough review by a panel of scientific experts. Based on the DOSITS website, this talk will provide background for the decision-making community on the characteristics of sound, underwater sound propagation, and appropriate measurement units. In addition, recent developments in the harmonization of sound modeling, measurement, and reporting will be discussed, highlighting the urgent need for consistent metrics across all underwater sound disciplines.

1:40

2pABa2. Underwater ears and the physiology of impacts: Comparative liability for hearing loss in sea turtles, birds, and mammals. Darlene R. Ketten (Biomedical Eng. and Otolology and Laryngology, Boston Univ. and Harvard Med. School, CMST, GPO Box 1987, Perth, Western Australia 6845, Australia, dketten@whoi.edu)

From human and laboratory animal studies, we have extensive knowledge about mechanisms of hearing loss from noise as well as aging, trauma, genetic conditions, and disease. By contrast, little is known about causes of hearing loss in wild species. Although there is great concern for anthropogenic acoustic impacts in marine species, especially for marine mammals, we are far from a clear understanding of the potential scope of impacts on the thousands of marine species. We have acquired significant data on hearing in pinnipeds, cetaceans, and fish, but far less is known about hearing and possible impacts in turtles and seabirds and how or if they suffer irreparable hearing loss. This talk will review what is known about hearing loss mechanisms from human and animal studies and based on the comparisons for land and marine ears, how the combined data can aid us to understand how hearing may be impaired in marine animals. It will also examine the current evidence for natural loss processes (presbycusis, disease, and trauma) in marine animals and implications for our ability to estimate and mitigate hearing loss from underwater sound.

2:00

2pABa3. Sound sources in the marine environment. Klaus Lucke (CMST, Curtin Univ., GPO Box U1987, Perth, WA 6845, Australia, klaus.lucke@wur.nl)

Regulation of underwater sound requires a good understanding of the sound emitted from various sound sources into the marine environment. Underwater sound sources can be subdivided into three main categories: geological, biological, and anthropogenic sources. While regulation of underwater sound self-evidently applies solely to anthropogenic sound, contributions from the other categories need to be taken into account when assessing the potential influence of man-made sound on the marine fauna. In addition to focusing on single offshore operations, cumulative noise exposure and cumulative stressors are relevant aspects for the regulation of underwater sound and need to be considered too. In this presentation, the main underwater sound contributors from each category will be identified, and the related sounds characterized and compared in relation to the overall marine sound energy budget. Important implications with regard to

physical parameters of sound will be briefly discussed. Current approaches to measuring, monitoring, and modeling underwater sound and how regulation can benefit from these different techniques will be reviewed.

2:20–3:20 Panel Discussion

3:20–3:40 Break

Contributed Papers

3:40

2pABa4. Implementing NOAA Fisheries' 2016 Marine Mammal Acoustic Guidance: Challenges and lessons learned. Amy R. Scholik-Schlomer (NOAA Fisheries Service, 1315 East-West Hwy, SSMC3, Rm. 13605, Silver Spring, MD 20910, amy.scholik@noaa.gov)

The National Oceanic and Atmospheric Administration's (NOAA) first comprehensive Guidance addressing the effects of noise on marine mammal hearing is intended for use by NOAA managers and applicants to better predict acoustic exposures that have the potential to trigger certain requirements under various statutes (e.g., U.S. Marine Mammal Protection Act; Endangered Species Act). The Guidance was developed by compiling, interpreting, and synthesizing scientific information on the effects of anthropogenic sound on marine mammal hearing. The Guidance's updated acoustic thresholds are more sophisticated than our previous thresholds. This added complexity is an important consideration for applicants who have formerly relied on more simple acoustic thresholds to evaluate potential impacts. Thus, the development of user-friendly tools is a fundamental issue for the regulatory community that is not often considered by most outside this group. As NOAA implements the Guidance, we have entered a new phase that consists of its own inherent issues and challenges associated with the practicality of employing more complex science to real-world applications. Throughout this process, NOAA has learned several valuable lessons, which will help improve the process of updating this document as well as drafting future guidance (e.g., marine mammal behavioral guidance; guidance for other protected species).

4:00

2pABa5. Cumulative sound exposure levels—Insights from seismic survey measurements. Bruce Martin, Jeff McDonnell (JASCO Appl. Sci., 32 Troop Ave., Ste. 202', Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com), and koen broker (Shell Global Solutions, Rijswijk, Netherlands)

The weighted cumulative sound exposure level from man-made noise sources is often recommended as a method of measuring possible injury to

marine life hearing. However, the behavior of this metric over large areas and its evolution in time is poorly documented in the scientific literature. Similarly, the differences in sound exposure levels as a result of changing the frequency weighting functions are only vaguely discussed. In this presentation, we provide insights into the behavior of the SEL metric based on measurements of several seismic surveys and simulations of seismic vessels passing a recording location. Based on the real-world measurements, we show how the range at which the surveys exceed the regulatory thresholds is highly dependent on the weighting functions and threshold values.

4:20

2pABa6. Taking into account uncertainties in environmental impact assessment of underwater anthropogenic noise. 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 management of underwater anthropogenic noise is becoming a significant component of marine policies. Rules to limit noise exposure in a way that marine animals are not adversely affected are expected to be set up in the future. However, modeling or measuring underwater noise levels (or other suitable metrics) can still be subject to a lot of uncertainties due to difficulty in estimation or measuring key parameters as source pressure levels and waveguide features. On the other hand, the audition threshold values inferred from bioacoustics studies may also lack of statistical robustness and are difficult to generalize. The combination of these two major sources of uncertainties may lead to misestimate the noise exposure risk and may hinder marine space planning efficiency. This work aims at developing a framework for impact studies while considering uncertainties on the acoustic metrics and confidence in the threshold values. It relies on probabilistic description of the acoustics pressure, as proposed in several recent studies. The formulation provides the interpretation of the results in terms of impact risk considering exposure time and source distance. Taking into account of the uncertainties becomes then a strong tool for decision support.

4:40–5:40 Panel Discussion

2pABb

Animal Bioacoustics: Data Management, Detection, Classification, and Localization

David K. Mellinger, Chair

*Coop. Inst. for Marine Resources Studies, Oregon State University, 2030 SE Marine Science Dr., Newport, OR 97365***Contributed Papers**

1:20

2pABb1. Organizing metadata from passive acoustic localizations of marine animals. Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., 5500 Campanile Dr, San Diego, CA 92182-7720, marie.roch@sdsu.edu), Philip Miller (Scripps Inst. of Oceanogr., San Diego, CA), Tyler A. Helble (Systems Ctr. Pacific, SPAWAR, San Diego, CA), Simone Baumann-Pickering, and Ana Širović (Scripps Inst. of Oceanogr., La Jolla, CA)

The Tethys system is a set of schemata designed to organize spatiotemporal data from detection, classification, and localization (DCL) tasks targeting sound producing marine animals. These metadata are based on the analysis of recordings collected through passive acoustic monitoring. Tethys is accompanied by a scientific workbench implementation of the schemata rules. The system is designed to promote the retention and organization of acoustic metadata in a manner that allows long-term retention of detailed DCL outputs. We present recent work on the localization schemata showing the ability to organize localizations from a large naval hydrophone range. The Tethys implementation permits these data to be analyzed in the context of Internet available data products containing oceanographic, atmospheric, and ephemeris data. Moving beyond simple detections, this addition of localizations to Tethys will allow even more powerful interpretation of the spatiotemporal occurrence of marine animals in the context of their ecology.

1:40

2pABb2. Advanced methods for passive acoustic detection, classification, and localization of marine mammals. David K. Mellinger, Yang Lu, Curtis Lending (Coop. Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu), Jonathan Klay (NOAA Pacific Marine Environ. Lab., Seattle, WA), David Moretti (Naval Undersea Warfare Ctr. (NUWC) Div., Wakefield, RI), Susan M. Jarvis (Naval Undersea Warfare Ctr. (NUWC) Div., Worcester, MA), Paul M. Baggenstoss (Naval Undersea Warfare Ctr. (NUWC) Div., Newport, RI), Stephen W. Martin (Environ. Dept., National Marine Mammal Foundation, San Diego, CA), Marie A. Roch, Christopher A. Marsh (Comput. Sci. Dept., San Diego State Univ., San Diego, CA), and Kaitlin E. Frasier (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

For effective long-term passive acoustic monitoring of large data sets, automated algorithms provide the ability to detect, classify, and locate (DCL) marine mammal vocalizations. Several DCL algorithms were developed and enhanced, with emphasis on methods robust to non-Gaussian and non-stationary noise sources. (1) A subspace model was developed to separate odontocete clicks from noise sounds. (2) A multi-class support vector machine was improved by resolving confusion among species' overlapping-frequency clicks, and a beaked whale buzz class developed. (3) For dolphin whistles, shape-related features, extractable automatically, were shown to carry species-specific information. (4) Equipment and site differences were discovered to affect Gaussian mixture model classifiers, and methods were developed to mitigate these differences. (5) A nearest-neighbor approach to detection association and 3D localization across multiple phones with

multiple arrivals was developed (and applied to beaked whales) using time-difference-of-arrival (TDOA) hyperbolic methods, retaining TDOAs with fewer than the usual three detections and using associations between a given phone's detections with nearest neighbors. (6) Minke whale "boing" frequency estimates were improved to differentiate individuals, and a kinematic tracking algorithm was developed. (7) A generalized-power-law detector for humpback whales was improved. (8) A software interface for detection was developed, then tested by sending data from *Ishmael* to a detection process in MATLAB.

2:00

2pABb3. Assessing the effects of noise masking and transmission loss on dolphin occupancy rates reported by echolocation click loggers deployed on the eastern Scottish coast. Kaitln Palmer (School of Biology, Univ. of St. Andrews, Sir Harold Mitchell Bldg., St Andrews, Fife KY16 9TH, United Kingdom, kp37@st-andrews.ac.uk), Kate L. Brookes (Marine Scotland Sci., Aberdeen, United Kingdom), and Luke Rendell (Univ. of St. Andrews, St. Andrews, United Kingdom)

C-PODs are commercially available echolocation click loggers used to monitor odontocete populations worldwide. Data from C-PODs have directly contributed to high-profile conservation efforts as well as provided major insights into cetacean behavior and habitat use. However, the "black-box" nature of the instruments poses a challenge to researchers seeking to validate data from these instruments. In this study, we simulate how changes in site-specific propagation conditions and ambient noise levels shift dolphin occupancy rates as reported by the C-POD. As part of the ECoMASS array, 10 calibrated continuous recorders (SM2Ms) were co-deployed with C-PODs in the North Sea. Transmission loss profiles, assumed dolphin source levels, and published C-POD performance metrics were combined to estimate the relationship between detection probability and ambient noise level at the 10 study sights. Bayesian models were then used to estimate dolphin occupancy rates with and without accounting for differences in detection probability. While absolute occupancy rates differed when detection probability was accounted for, relative trends in occupancy were generally consistent within the two models. These data suggest that, within the scope of the ECoMASS array, relative occupancy rates are somewhat robust to differences in transmission loss and ambient noise levels throughout the survey period and location.

2:20

2pABb4. Variability in ground-truth data sets and the performance of two automated detectors for Antarctic blue whale calls in different soundscape conditions. Emmanuelle C. Leroy (Laboratoire GeoSci. Ocean, Univ. of Brest, IUEM Technopole Brest Iroise, Rue Dumont d'Urville, Plouzané 29280, France, emmanuelle.leroy@univ-brest.fr), Karolin Thomisch, and Ilse Van Opzeeland (Ocean Acoust. Lab, Alfred Wegener Institut, Bremerhaven, Germany)

Automated detectors are important tools for processing large passive acoustic databases. Assessing the performance of a given method can be challenging and needs to be interpreted in the light of the overall purpose of analysis. Performance evaluation often involves comparison between the

detector output and a ground-truth data set, which often involves manual analyses of the data. Such analyses may be subjective depending on, e.g., interfering background noise conditions. In this study, we investigated the variability between two analysts in the detection of Antarctic blue whale Z-calls (*Balaenoptera musculus intermedia*), as well as the intra-analyst variability, in order to understand how this variability impacts the creation of a ground-truth and the assessment of detector performances. Analyses were conducted on two test datasets reflecting two basins and different situations of call abundance and background noise conditions. Using a ground-truth based on combined results of both analysts, we evaluated the performances of two automated detectors, one using spectrogram correlation and the other using a subspace-detection strategy. This evaluation allows understanding how recording sites, vocal activity, and interfering sounds affect the detector performances and highlights the advantages and limitations of each of the methods, and the possible solutions to overcome the main limitations.

2:40

2pABb5. Automatic classification of humpback whale social calls. Irina Tolkova (Appl. Mathematics, Univ. of Washington, Durham, NH), Lisa Bauer (Comput. Sci., Johns Hopkins, 401 N Coquillard Dr., South Bend, IN 46617, lbauer6@jhu.edu), Antonella Wilby, Ryan Kastner (Univ. of California San Diego, San Diego, CA), and Kerri Seger (Univ. of New Hampshire, La Jolla, CA)

Acoustic methods are an established technique to monitor marine mammal populations and behavior, but developments in computer science can expand the current capabilities. A central aim of these methods is the automated detection and classification of marine mammal vocalizations. While many studies have applied bioacoustic methods to cetacean calls, there has been limited success with humpback whale (*Megaptera novaeangliae*) social call classification, which has largely remained a manual task in the bioacoustics community. In this project, we automated this process by analyzing spectrograms of calls using PCA-based and connected-component-based methods, and derived features from relative power in the frequency bins of these spectrograms. These features were used to train and test a supervised Hidden Markov Model (HMM) algorithm to investigate classification feasibility. We varied the number of features used in this analysis by varying the sizes of frequency bins. Generally, we saw an increase in precision, recall, and accuracy for all three classified groups, across the individual data sets, as the number of features decreased. We will present the classification rates of our algorithm across multiple model parameters. Since this method is not specific to humpback whale vocalizations, we hope it will prove useful in other acoustic applications.

3:00–3:20 Break

3:20

2pABb6. Cepstral analysis of vocal tract resonance from Tree Swallow (*Tachycineta bicolor*) songs during learning. Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com)

Bird vocalizations range in spectral composition from pure-tone whistles through resonant bugles to noisy squawks. Birds differ from mammals in the vibratory source of their vocalizations. Individuals in both groups use muscular control of the resonances of their vocal tracts to affect the resulting acoustic signal. Sensorimotor learning about this process plays an important role in the learning of both human speech and avian song. It is hypothesized that vocal tract resonances, as measured by cepstrum-based formant detectors, will show more variation both among and within calls early in the learning process. This should be particularly true for birds whose songs consist of rapidly frequency-modulated pure tones. Field recordings of multiple wild tree swallows (*Tachycineta bicolor*) are analyzed over the course of the breeding season. The consistency of vocal tract resonances are compared among birds of different ages and between newly- and previously-learned song units.

3:40

2pABb7. Estimating geo-position and depth of echo-locating beaked whales using an array of drifting recorders. Jay Barlow, Emily T. Griffiths (Marine Mammal and Turtle Div., NOAA-NMFS-SWFSC, 8901 La Jolla Shores Dr., La Jolla, CA 92037, jay.barlow@noaa.gov), and Holger Klinck (BioAcoust. Res. Program, Cornell Lab of Ornithology, Ithaca, NY)

An array of 8 drifting recorders were deployed in the Catalina Basin off Southern California to localize beaked whales. The drifting recorders with hydrophone pairs at 90-135 m were deployed along two parallel lines with ~1 km separation between recorders. The array was re-deployed daily at approximately the same location to maintain this array spacing. Cuvier's beaked whales (*Ziphius cavirostris*) were detected on 26 occasions from their distinctive echo-location clicks. On 8 of these occasions, direct-path and surface-reflected signals were received on four or more drifting recorders, which allowed us to estimate location and depth of the whales. Average array tilt during these events was less than 0.2°, and always less than 0.6°. The same echolocation clicks were seldom received on more than two recorders, so we could not use methods that require TDOA measurements between recorders. We develop a novel method of 3-D localization using the vertical bearing angles estimated from the direct- and surface-reflected signals and use optimization methods to find the unique location and depth at which these angles converged. Detection ranges varied from 1.0 to 3.7 km (mean=2.0, sd=0.65), and depths of vocalizing animals varied from 696-1150 m (mean=948, sd=152).

4:00

2pABb8. Analysis of marine mammal bearing tracks from two-hydrophone recordings made with a glider. Elizabeth T. Küsel and Martin Siderius (Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, kusele@alum.rpi.edu)

An underwater glider fitted with two hydrophones recorded approximately 19 hours of data during an opportunistic sea experiment in the summer of 2014. The acoustic data were collected with a sampling frequency of 96 kHz and 16-bit resolution in deep waters off the western coast of the island of Sardinia, Mediterranean Sea. Detection and classification of sounds by a trained human analyst indicated the presence of sperm whale (*Physeter macrocephalus*) regular clicks as well as dolphin clicks and whistles. A period of 90 min during which the glider did not surface, and which contained extensive sperm whale clicking activity was chosen for analysis. Cross-correlation of the data from both channels allowed the estimation of the direction (bearing) of clicks, and realization of animal tracks. Several bearing tracks were observed through this analysis, closely following the oscillatory pattern of the glider's heading, suggesting that such information has the potential to break the left-right ambiguity of the bearing estimates. Results from the bearing tracking analysis, including accuracy and performance, will be shown followed by a discussion on how they can aid in population density estimation studies.

4:20

2pABb9. A comparison of whale tracking algorithms in the Indian Ocean using the Comprehensive Test Ban Treaty Hydrophone Data. David A. Lechner (Mech. Eng., The Catholic Univ. of America, 9404 Bethany Pl., Washington, DC 20064, 66Lechner@cua.edu), Shane Guan (Mech. Eng., The Catholic Univ. of America, Silver Spring, MD), and Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This paper will present results comparing several geo-location and tracking algorithms applied to whale signals as monitored by the Comprehensive Test Ban Treaty (CTBT) sensor network. The performance of three tracking algorithms are compared using several hours of acoustic data collected off Cape Leeuwin, Australia, containing the apparent broadcast call of a blue whale (*Balaenoptera musculus*). The first approach used cross-correlation in

2p MON. PM

time on the entire signal, while the second used a smaller time sample that covers the leading or trailing edge of the signal and creates a motion vector from the pair. The third approach used the Cross-Ambiguity-Function Mapping technique to generate 2-D energy maps on a geographic plot. Accuracy and energy detection algorithms based on blind-processing of the data are compared.

4:40

2pABb10. Bat population censusing with passive acoustics. Laura Kloeppe, Yanqing Fu, and Joel Ralston (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, lkloeppe@saintmarys.edu)

Passive acoustic monitoring is a widely used method to identify bat species and determine spatial and temporal activity patterns. One area where acoustic methods have not yet been successfully applied, however, is in determining population counts, especially from roosts. Typically, most roost counts are obtained with thermal imagery that may be prohibitively expensive for many natural resource managers or require complex computer programming. Here, we demonstrate a new acoustic technique to estimate population size of Brazilian free-tailed bats (*Tadarida brasiliensis*) emerging from large cave colonies. Data were acquired across multiple nights and at 9 cave locations with different roost structures and flight behavior profiles. We used a single microphone to monitor echolocation activity and simultaneously recorded the emerging bats with thermal video. Bat abundance counts were determined from a single video frame analysis (every 10 s) and were compared to different acoustic energy measures of a 1-s long acoustic sequence recorded at the time of the analyzed video frame. For most cave locations, linear regression models successfully predicted bat emergence count based on acoustic intensity of the emerging stream. Here, we describe our method and report on its application for counting bats from different roost locations.

5:00

2pABb11. Evaluating autonomous underwater vehicles as platforms for animal population density estimation. Danielle Harris (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, The Observatory, Buchanan Gardens, St. Andrews KY16 9LZ, United Kingdom, dh17@st-andrews.ac.uk), Selene Fregosi (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR), Holger Klinck (BioAcoust. Res. Program, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY), David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR), Jay Barlow (Marine Mammal and Turtle Div., NOAA Southwest Fisheries Sci. Ctr., La Jolla, CA), and Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, United Kingdom)

AFFOGATO (A Framework For Ocean Glider-based Acoustic density estimation) is a multi-year project (2015–2018) funded by the Office of Naval Research. Its main goal is to investigate the utility of slow-moving marine vehicles, particularly ocean gliders and profiling floats, for animal density or abundance estimation, using the passive acoustic data that these vehicles can collect. In this presentation, we will (1) provide a project overview and (2) share results from the initial stages of the project. As part of one task, existing deployments in the Gulf of Alaska, Hawaii, and the Mariana Islands have been used to investigate the capability of gliders to adhere to planned survey tracks. Simulations were also conducted to assess whether realized glider survey track lines could produce unbiased density estimates using two hypothetical animal distributions and assuming so-called design-based analysis methods (the standard, and also simplest, approach). Five deployments were assessed and deviances of up to 20 km from the planned survey track line were found. Under the specific simulated conditions, density estimates showed biases up to 9%. Next stages of the project will also be discussed, including ongoing work to estimate the probability of detecting different cetacean species using AUVs.

2pAO

Acoustical Oceanography: Session in Honor of David Farmer II

Andone C. Lavery, Cochair

Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 98 Water Street, MS 11, Bigelow 211, Woods Hole, MA 02536

Grant B. Deane, Cochair

Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St., La Jolla, CA 92093-0238

Tim Leighton, Cochair

*Institute of Sound and Vibration Research, University of Southampton, Southampton, United Kingdom***Invited Paper**

1:20

2pAO1. Subtleties in the acoustics of marine sediments. Michael J. Buckingham (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

Based on the pioneering work of David Farmer, it is now well recognized that the layer immediately beneath the sea surface is a highly dynamic, two-phase region, where bubbles created by wave breaking form an upward refracting sound speed profile that acts as an acoustic waveguide. The bottom boundary, although less dynamic than the sea surface, possesses its own unique complexities that are no less challenging to understand than those of the near-surface bubble layer. For instance, a growing body of experimental evidence indicates that the acoustic attenuation in a marine sediment obeys a frequency power law, extending over a wide bandwidth, in which the exponent takes a value close to unity. A long-standing problem has been to identify the frequency dispersion in the sound speed associated with such a power-law attenuation. Several solutions to this problem have been proposed over recent decades but are in fact unphysical in that they fail to obey the Kramers-Kronig dispersion relations. An alternative solution that has recently been developed, which overcomes the previous difficulties, contains a number of subtleties that, it is hoped, will appeal to David Farmer's keen sense of scientific curiosity. [Research supported by ONR.]

Contributed Papers

1:40

2pAO2. A method of estimating the value of *in situ* surface tension on a bubble wall. Mengyang Zhu, Tim Leighton (Inst. of Sound and Vib. Res., Eng. and the Environment, Univ. of Southampton, University Rd., Southampton, Hampshire SO17 1BJ, United Kingdom, M.Zhu@soton.ac.uk), and Peter Birkin (Chemistry, Univ. of Southampton, Southampton, Hampshire, United Kingdom)

The surface tension of a liquid is an important parameter for estimating and analyzing the processes that happen on the air/liquid interface, such as the air/sea gas exchange. Current methods of measuring surface tension concentrate on measuring its value at the top flat air/liquid interface. However, in cases where bubbles mediate oceanic processes (such as their contributions to air-to-sea transfers of mass, energy, and momentum), the value of surface tension that is needed (e.g., for placement in models of the evolution and persistence of sub-surface bubble clouds) is the instantaneous value on the bubble wall, as it moves through the ocean and potentially collects surface-active species onto the bubble wall. This paper outlines a method of estimating the value of this *in situ* surface tension, by insonifying a bubble and observing the onset of Faraday waves on a bubble wall. This new method was compared with a traditional ring method in various scenarios.

2:00

2pAO3. Experimental observations of acoustic backscattering from spherical and wobbly bubbles. Alexandra M. Padilla, Kevin M. Rychert, and Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, 24 Colovos Rd., Durham, NH 03824, apadilla@ccom.unh.edu)

Methane bubbles released from the seafloor transport gas through the water column to the atmosphere. Direct or optical methods via underwater vehicles are often used for quantifying methane gas flux in the water column, however these methods are time consuming and expensive. Acoustic measurements, using split-beam and multibeam echo sounders that are readily available on most sea vessels, provide a more efficient method for determining methane gas flux. These acoustic methods typically convert acoustic backscatter measurements of bubbles to bubble size using analytical models of bubble target strength. These models assume that bubbles have uniform shape; however it has been shown that bubbles with a radius greater than 1 mm, which have large Eötvös and/or large Reynolds number, are non-spherical. To investigate the error associated with assuming large bubbles are spherical, a 6 m deep tank experiment was conducted to compare calibrated target strength measurements of both small spherical and large wobbly bubbles to existing acoustic scattering models. Bubble sizes observed in this experiment ranged from a fraction of 1 mm to 6 mm in radius. This experiment used a broad range of frequencies (10-300 kHz) to cover typical echo sounder frequencies utilized in field measurements of natural methane seeps.

2pAO4. Low-frequency active acoustic response of underwater bubble plumes. Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu), Zel J. Hurewitz (Dept. of Phys., The Univ. of Texas at Austin, Austin, TX), and Paul M. Abkowitz (George W Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Active, low frequency acoustics may provide a method of locating and identifying methane bubble plumes over a large area by exciting radial and column resonances. In this work, a proof of concept experiment was conducted by measuring the active, low frequency response of an air bubble plume at the Lake Travis Test Station at the University of Texas at Austin. Several flow rates, bubble sizes, and plume widths were investigated. The test consisted of insonifying the plumes at two different standoff distances using a low-frequency chirp from 300-2500 Hz. The plume response was much longer in time than originally expected due to resonant ringing. The long-time response was modeled with a finite element model, which predicted the column resonance. This behavior may be able to be exploited for long-range bubble plume identification. [Work supported by ExxonMobil.]

2pAO5. Efficacy of up and down-chirps for two-pulse sonar techniques. Nikhil Mistry, Paul White, and Tim Leighton (Inst. of Sound & Vib. Res., Southampton Univ., Bldg. 13, Rm. 3049, University Rd., Southampton, Hampshire SO17 1BJ, United Kingdom, nm6g09@soton.ac.uk)

Twin-Inverted Pulse Sonar (TWIPS) and Biased Pulse Summation Sonar (BiaPSS) were inspired by a video of dolphins using bubble nets to hunt. There are a number of animals that have been observed to use the chirp (a sweep in sine waves of shifting frequency) in echolocation. This talk will present the difference in efficacy of the sonar techniques, mentioned above, when using upwards and downwards sweeping chirps as sonar signals. These techniques have been simulated and experimentally tested with a number of bubble size distributions (BSD). This includes simulations using the results of BSD measurements by Farmer and Vagle (1989). Further, another two-pulse technique will be outlined, proposed as a means to overcome the limitation on detection range, owing to the inter-pulse delay in TWIPS and BiaPSS.

Invited Papers

3:00

2pAO6. An acoustic study of sea ice behavior in a shallow, Arctic bay. Oskar Glowacki (Inst. of Geophys., Polish Acad. of Sci., Ksiecica Janusza 64/413, Warsaw 01-452, Poland, oglowacki@igf.edu.pl)

Recent acceleration of sea ice decline observed in the Arctic Ocean draws attention to environmental factors driving this phenomenon. One of the main conclusions is a growing need for better understanding of sea ice drift, deformation, and fracturing. In response to that call, several ambient noise recordings were carried out in the coastal zone of Hornsund Fjord, Spitsbergen, in spring 2015 to study underwater acoustic signatures of sea ice behavior. The noise levels varied significantly with sea ice type and intensity of external forces. Low-frequency signatures were strongly related to tidal cycle, which manifested in much higher SPL values at low water. Compacted ice cover is periodically deformed and crushed, representing a significant contribution to the ambient noise field in the study site. Average noise levels at frequencies above 1 kHz are, in turn, considerably higher in front of marine-terminating glacier than in the neighboring, non-glacial bay. These differences, expanding with the rise of water temperature, are associated with melting of the ice cliff and generally unaffected by the presence of sea ice. [Work funded by the Polish National Science Centre, Grant No. 2013/11/N/ST10/01729.]

3:20–3:40 Break

3:40

2pAO7. The masking of beluga whale (*Delphinapterus leucas*) sounds by icebreaker noise in the Arctic. Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au)

Beluga whales are an Arctic and subarctic cetacean, with an overall “near threatened” conservation status, yet some populations are considered endangered. Apart from threats such as whaling, predation, contamination, and pathogens, underwater noise is of increasing concern. In the early 1990s, Fisheries & Oceans Canada started to fund research on underwater noise emitted by icebreakers and its bioacoustic impacts. In collaboration with the Vancouver Aquarium, beluga whales were trained for masked hearing experiments. Apart from measuring pure-tone audiograms in quiet conditions, animals were trained to listen for beluga vocalizations in different types of noise, including artificially created white noise, naturally occurring thermal ice-cracking noise, and an icebreaker’s propeller cavitation and bubbler system noise. Based on this data, software models for masking in beluga whales were developed. More than 20 years later, this dataset remains the only one on masking in cetaceans using both complex signals (actual vocalizations) and complex noise (actual recordings of Arctic ambient and anthropogenic noise)—highlighting both Dave Farmer’s foresight as well as perhaps his lesser-known escapades into marine mammal bioacoustics.

4:00

2pAO8. The upper ocean ambient sound field as a tool to address significant scientific and societal questions. Svein Vagle (Fisheries and Oceans Canada, Inst. of Ocean Sci., 9850 West Saanich Rd., Sydney, BC V8L4B2, Canada, Svein.Vagle@dfo-mpo.gc.ca)

David Farmer’s keen interest in understanding the dynamics of the upper-ocean and air-sea interaction convinced him that studying and using the naturally occurring high-frequency oceanic sound field would give additional insight into these processes. Breaking surface waves are important to ocean dynamics and were believed to be a significant source of the observed sound field. However, direct measurements were lacking. In the mid-1980s and onwards, David Farmer and his students developed a range of new observational techniques and instrumentation which would significantly improve our understanding of the sound field itself and how it relates to

parameters such as wind speed and wave conditions, and as a tool to understanding the more fundamental physical processes of the upper ocean. All this research resulted in significant progress in our understanding of wave breaking, air-sea interaction, air-sea gas transfer, and indirectly to upper-ocean bubble distributions and dynamics, ice generated sound, and the role of anthropogenic noise on marine fauna. Here, we review key outcomes of this research and discuss how several components of Farmer et al.'s work now is being used to address significant societal issues with regard to the impacts of increasing levels of man-made underwater noise on marine life.

4:20

2pAO9. Passive and active acoustical studies of ocean surface waves. Li Ding (Vitech Res. and Consulting, 6280 Doulton Ave., Richmond, BC V7C4Y4, Canada, lding2011@gmail.com)

This paper reviews previous work with Professor David Farmer, on the use of acoustical techniques to observe and measure ocean surface waves. In passive acoustics, breaking surface waves in the open ocean were observed with a hydrophone array deployed close to the surface to track individual breaking events. The spatial and temporal statistics of braking events, such as velocity and breaking probability, were determined and compared with simultaneously measured directional wave spectra. The comparisons suggest that wave breaking occurs at multiple scales and that the mean scale of breaking is substantially smaller than the associated with the dominant wind wave component. In an active acoustical study, an incoherent bistatic sonar mounted on the seafloor was used to measure currents close to the ocean surface and within the crests of large, steep waves. Individual estimates of the currents at, and close to, the surface were made with sufficient temporal resolution to identify kinematics in the crests of large waves. Observations acquired in the North Sea are examined to evaluate both the potential merits and limitations of the measurement approach. The observations lead to some conclusions regarding wave kinematics during a storm in which the wind speed reached $\sim 17 \text{ m s}^{-1}$.

Contributed Papers

4:40

2pAO10. What can we learn from breaking wave noise? Grant B. Deane (Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St., La Jolla, CA 92093-0238, gdeane@ucsd.edu)

Breaking waves are an important process at the air-sea interface: they limit the growth of waves, transfer momentum between the atmosphere and ocean, generate marine aerosols, change ocean albedo, and enhance the transport of greenhouse gases across the air-sea interface. However, they are a challenging phenomenon to study in the field and much is yet to be learnt about the transient, two phase flow inside whitecaps. Acoustical oceanography has much to offer as a remote sensing tool for studying wave breaking, and has been exploited to great effect by David Farmer and his colleagues over the past 3 decades. I will cover some of the highlights of this fascinating subject, including recent developments to study air entrainment and turbulence in breaking waves using radiated wave noise. [Work supported by ONR, Ocean Acoustics Division, and NSF.]

5:00

2pAO11. Insights from acoustical oceanography: A personal assessment. David Farmer (Inst. of Ocean Sci., Vancouver, BC, Canada, farmer.davidm@gmail.com)

New ways of looking at the natural environment often lead to new insights. It is true of the ocean and especially so in the rapidly developing field of acoustical oceanography. This will be illustrated with examples drawn from personal experience over the past 3-4 decades, having application to a range of phenomena from coastal processes to air-sea interaction, from stratified flow over topography to sea ice behavior. Much is owed to the Acoustical Society's welcoming approach to oceanographers, to the skill and enthusiasm of our students, colleagues, and staff, and to the invaluable support of our sponsors.

5:20–5:40 Panel Discussion

2p MON. PM

2pBA**Biomedical Acoustics and Physical Acoustics: Beamforming and Image Reconstruction**

Martin D. Verweij, Cochair

Acoustical Wavefield Imaging, Delft University of Technology, Lorentzweg 1, Delft 2628CJ, Netherlands

Hendrik J. Vos, Cochair

*Biomedical Engineering, Erasmus MC, Rotterdam, Netherlands***Chair's Introduction—1:15*****Invited Papers*****1:20****2pBA1. Fast compressive pulse-echo ultrasound imaging using random incident sound fields.** Martin F. Schiffner and Georg Schmitz (Medical Eng., Ruhr-Univ. Bochum, Universitätsstr. 150, Bochum 44801, Germany, martin.schiffner@rub.de)

In fast pulse-echo ultrasound imaging (UI), the image quality is traded off against the image acquisition rate by reducing the number of sequential wave emissions per image. To alleviate this tradeoff, the concept of compressed sensing (CS) was proposed by the authors in previous studies. CS regularizes the linear inverse scattering problem (ISP) associated with fast pulse-echo UI by postulating the existence of a nearly-sparse representation of the object to be imaged. This representation is obtained by a known linear transform, e.g., the Fourier or a wavelet transform. A central degree of freedom in the regularized ISP is the choice of the incident sound fields. Previous studies focused exclusively on steered plane waves. In this study, we investigate the usage of *random incident sound fields* to improve the relevant mathematical properties of the scattering operator governing the linear ISP. These sound fields are synthesized by a linear transducer array whose physical elements are excited applying combinations of random time delays and random apodization weights. Using simulated and experimentally obtained radio frequency signals, we demonstrate that these sound fields significantly reduce the recovery errors and improve the rate of convergence for low signal-to-noise ratios.

1:40**2pBA2. Model-based image reconstruction for medical ultrasound.** Pieter Kruizinga (Biomedical Eng., Erasmus MC, Westzeedijk 353, BME Ee3202, Rotterdam 3015AA, Netherlands, p.kruizinga@erasmusmc.nl), Pim van der Meulen (Circuits and Systems, Delft Univ. of Technol., Delft, Netherlands), Frits Mastik, Nico de Jong, Johannes G. Bosch (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), and Geert Leus (Circuits and Systems, Delft Univ. of Technol., Delft, Netherlands)

Most techniques that are used to reconstruct images from raw ultrasound signals are based on pre-defined geometrical processing. This type of image reconstruction typically has a low computational complexity and allows for real-time visualization. Since these techniques do not account for situation-specific parameters such as transducer characteristics and medium in-homogeneities, they cannot make proper use of the information that is contained in the raw ultrasound signals. In this paper, we explore the possibility of reconstructing images that best explain the measured ultrasound signals given the full ultrasound propagation model including all parameters. We build this model by measuring the spatiotemporal impulse response of the imaging transducer and, using the angular spectrum approach, estimate the ultrasound signal as it would originate from each individual image pixel position. An iterative search for the pixel combination that best explains the recorded signals provides the final image. We discuss the details of this model, provide experimental proof that this reconstruction allows for improved image quality, and extend our ideas to other imaging schemes such as compressive imaging.

2:00**2pBA3. Beamforming methods for large aperture imaging.** Gregg Trahey, Nick Bottenus (Biomedical Eng., Duke Univ., 136 Hudson Hall, Box 90281, Durham, NC 27708, gregg.trahey@duke.edu), and Gianmarco Pinton (Biomedical Eng., Univ. of North Carolina, Chapel Hill, NC)

Maintaining image quality at large tissue depths remains a clinically significant and unmet challenge for ultrasonic scanners. For tissue structures beyond 10 cm, commonly encountered in obstetric and abdominal scans, diffraction and propagation through tissue can limit azimuthal resolution to be larger than 5 mm and elevation resolution can exceed a few centimeters, making the evaluation of sub-centimeter fetal anatomical features or renal or hepatic lesions very difficult. We describe simulation and ex vivo human tissue studies which evaluate the image quality achievable with large aperture arrays and with associated beamforming methods. We imaged through human abdominal tissue layers and synthetically formed very large coherent apertures (2 cm X 10 cm) apertures. We also performed matched simulations using Visible Human Project-derived tissue models and full-wave simulation code. Using both datasets, we

assessed : 1) the image quality improvements attainable with large arrays, 2) the factors degrading the image quality of deep-lying tissues, and 3) the beamforming methods best suited for large array imaging of deep tissues. Our results indicate that large improvements in resolution are obtainable for deep-lying tissues when imaging with large arrays. The major source of tissue-induced image degradation was observed to be clutter due to reverberation and beamforming limitations, rather than phase errors. Coherence-based beamforming methods were seen to be especially applicable in large array imaging.

2:20

2pBA4. Reverberation suppression and enhanced sensitivity by coherence-based beamforming. Jeremy J. Dahl (Radiology, Stanford Univ., 3155 Porter Dr., Palo Alto, CA 94304, jeremy.dahl@stanford.edu)

Diffuse acoustic reverberation is often present in poor quality ultrasound images and can mask organs and anatomical structure. In addition, imaging methods such as blood flow and targeted microbubble imaging can be problematic in the presence of reverberation and thermal noise. We present a beamforming method that is based on the spatial coherence of backscattered ultrasound waves. The method differentiates signal from noise based on the spatial coherence of the signal and small spatial differences (or lags), and is therefore called the Short-Lag Spatial Coherence (SLSC) beamformer. Because diffuse reverberation and thermal noise are spatially incoherent, they can easily be distinguished from tissue, blood, and other signals of interest. We present SLSC beamforming and its applications to cardiac and other imaging targets, tissue harmonic imaging, flow imaging, and molecular imaging. We show that the technique improves the visibility of organ structures and the sensitivity of flow and molecular imaging targets by suppression of noise signals. We demonstrate a real-time SLSC beamforming prototype system that achieves upwards of 35 fps in cardiac imaging and 50 fps in molecular imaging. The system demonstrates high-quality, stable images of *in vivo* organs and targets using the SLSC beamformer.

2:40

2pBA5. Ultrafast ultrasound imaging temporal resolution enhancement with filtered delay multiply and sum (FDMAS) beamforming. Asraf Moubark, Zainab Alomari, Sevan Harput, David Cowell, and Steven Freear (School of Electron. Eng., Univ. of Leeds, Leeds LS2 9JT, United Kingdom, s.freear@leeds.ac.uk)

FDMAS beamforming technique have been employed with low number of steering angles and smaller lateral steps in order to improve the image quality for better cyst classification. Taking advantage of the autocorrelation process in FDMAS, lateral steps were reduced in order to calculate the time delay for the RF signal more accurately. The new beamforming technique has been tested on CIRS phantoms experimentally with the ultrasound array research platform version 2 (UARP II) using a 3-8 MHz 128 element clinical transducer. The point spread function (PSF) main lobes lateral resolution measured at 20 dB shows an improvement of 65.8% for lateral steps λ to $\lambda/5$. Meanwhile, the contrast ratio (CR) obtained for an anechoic cyst size 1 mm in diameter located at 15 mm depth with lateral steps of λ and $\lambda/5$ are -11.7 dB and -18.36 dB, respectively. The contrast to noise ratio (CNR) also shows improvement of 17.6% for the same lateral steps reduction. In conclusion, reducing lateral steps in FDMAS beamforming technique with low number of steering angles outperforms DAS with a high number of steering angles in laboratory experiments by narrowing its main lobes and increasing the image contrast thus improve the temporal resolution.

3:00

2pBA6. An adaptive mirage. Alfonso Rodriguez-Molares (Circulation and Medical Imaging, Norwegian Univ. of Sci. and Technol., Det medisinske fakultet, Institutt for sirkulasjon og bildediagnostikk, Postboks 8905, Trondheim 7491, Norway, alfonso.r.molares@ntnu.no), Ole Marius H. Rindal (Univeristy of Oslo, Oslo, Norway), Ali Fatemi (Circulation and Medical Imaging, Norwegian Univ. of Sci. and Technol., Trondheim, Norway), and Andreas Austeng (Univ. of Oslo, Oslo, Norway)

We are in the middle of a Cambrian explosion. Software beamforming has redefined what can be done with the signal. As a consequence, our field has become flooded with adaptive beamforming (AB) algorithms, methods that by clever manipulation of channel data have exceeded our wildest expectations for the maximum achievable contrast and resolution. Or have they? If we define image quality in terms of the contrast ratio (CR) and the full-width half-maximum (FWHM), there is another way of getting unprecedented image quality. Dynamic range stretching, the kind of stretching one gets from squaring the beamformed signal amplitude, will also produce higher CR and smaller FWHM. If AB alters the output dynamic range, then the reported CR and FWHM are invalid. No tools are available yet for researchers and reviewers to check this. Here we address this problem. We propose a phantom to measure the dynamic range of AB. The phantom includes a speckle gradient band similar to those used in the calibration of monitors. The phantom allows us to confirm that AB algorithms can alter the dynamic range of the signal and produce incorrect CR and FWHM values. But it also makes it possible to compensate for that alteration and calibrate the algorithms. After calibration AB still results in higher image quality than delay-and-sum, but the metrics are more reasonable. A debate must be opened on the significance of AB algorithms. The metrics used to assess image quality must be revised. Otherwise, we risk to walk in circles, tricked by an illusion.

3:20–3:40 Break

3:40

2pBA7. A Fresnel-inspired approach for steering and focusing of pulsed transmit beams by matrix array transducers. Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628CJ, Netherlands, m.d.verweij@tudelft.nl), Michiel A. Pertijs (Electron. Instrumentation, Delft Univ. of Technol., Delft, Netherlands), Jos de Wit, Fabian Fool (Acoust. Wavefield Imaging, Delft Univ. of Technol., Delft, Netherlands), Hendrik J. Vos (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), and Nico de Jong (Acoust. Wavefield Imaging, Delft Univ. of Technol., Delft, Netherlands)

Matrix ultrasound transducers for medical diagnostic purposes are commercially available for a decade. A typical matrix transducer contains 1000+ elements, with a trend towards more and smaller elements. This number renders direct connection of each individual element to an ultrasound machine impractical. Consequently, it is cumbersome to employ traditional focusing and beamforming approaches that are based on transmit and receive signals having an individual time delay for each element. To reduce cable count

during receive, one approach is to apply sub-arrays that locally combine the element signals using programmable delay-and-sum hardware, resulting in reduction by a factor 10. In transmit, achieving cable count reduction while keeping focusing and steering capabilities turns problematic once it becomes impossible to locally equip each element with its own high voltage pulser. To overcome this bottleneck for decreasing element size, here we present a Fresnel-inspired hardware and beam forming approach that is based on transmit pulses consisting of several periods of an oscillating waveform. These will be derived from one oscillating high voltage signal by using local switching and timing hardware. To demonstrate the feasibilities of our approach, we will show beam profiles and images for a miniature matrix transducer that we are currently developing.

Contributed Papers

4:00

2pBA8. *In vivo* measurements of muscle elasticity applying shear waves excited with focused ultrasound. Timofey Krit, Valeriy Andreev (Phys., Moscow State Univ., Leninskie Gory, Bldg. 1/2, Moscow, Moscow 119991, Russian Federation, timofey@acs366.phys.msu.ru), Igor Demin (Lobachevsky State Univ. of Nizhny Novgorod, Nizhny Novgorod, Russian Federation), Pavel Rykhtik, and Elena Ryabova (Federal Inst. of Health «Privolzhsky Regional Medical Ctr. Federal Medical-Biological Agency of Russia», Nizhny Novgorod, Russian Federation)

The common algorithm of shear waves excitation for diagnostical ultrasonic devices was modified for measurements in muscles. We measured the speed of shear waves, excited by a focused ultrasound at a frequency of 5 MHz in the muscles of the volunteers. Siemens Acuson S2000 was used for *in vivo* measurements. The suggested algorithm was tested on the muscle mimicking phantoms. The values of shear wave velocities in the same areas of studied phantoms at the same angles measured with Siemens Acuson S2000 system corresponded to the values obtained by Verasonics, where the region of shear wave excitation had a form of “blade” of thickness less than 0.5 mm, length and width of 1.5-2 mm. Due to this form of the region, the excited shear wave has propagated codirectional with the long side of the ultrasonic medical probe. Thus, the direction of propagation of the shear wave with respect to the phantom fibers, became dependent on the position of the probe. [The reported study was funded by RFBR and Moscow city Government according to the research project № 15-32-70016 «mol_a_mos», by RFBR according to the research project № 16-02-00719 a, and by Program for Sponsorship of Leading Scientific Schools (Grant NSh-7062.2016.2).]

4:20

2pBA9. Non-invasive carotid artery elastography using multi-element synthetic aperture and plane wave imaging: Phantom and *in vivo* evaluation. Rohit Nayak (Elec. and Comput. Eng., Univ. of Rochester, 205 Conant Rd., Apartment B, Rochester, NY 14623, rohitnayak@rochester.edu), Giovanni Schifitto (Dept. of Neurology, Univ. of Rochester Medical Ctr., Rochester, NY), and Marvin M. Dooley (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

Vascular elastography can visualize the strain distribution in the carotid artery, which governs plaque rupture. In this study, we hypothesize that multi-element synthetic aperture (MSA) imaging, which produces divergent transmit beams can overcome the grating lobes issues associated with compounded plane wave (CPW) imaging, and produce more reliable strain elastograms. To corroborate this hypothesis, we conducted phantom and *in vivo* studies using both the techniques, and determined the most optimal imaging configuration for carotid elastography. The phantom studies were conducted using cryogel vessel phantoms. We validated the phantom study results *in vivo*, on healthy volunteers. These studies were performed using a commercially ultrasound scanner (Sonix RP, Ultrasonix Medical Corp., Richmond, BC, Canada), operating at a transmit frequency of 5 MHz. The phantom results demonstrated that the plaque was visible in elastograms from both techniques; however, MSA elastograms had fewer artifacts, with a 12 dB improvement in elastographic contrast to noise ratio relative to CPW imaging. Further, the results from the *in vivo* study agreed with the phantom results. These results suggest that MSA imaging can produce useful strain elastograms. Our future work will involve further development and more *in vivo* validation.

4:40

2pBA10. Cramer Rao lower bound for two-dimensional elastography. Prashant Verma (Dept. of Phys. and Astronomy, Univ. of Rochester, 60 Crittenden Blvd. Apt. 230, Rochester, NY 14620, prashant.v.iitkgp@gmail.com) and Marvin M. Dooley (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

In this study, we present a theoretical framework for characterizing the performance of two-dimensional displacement and strain estimators. Specifically, we derived the Cramer-Rao lower bound for axial and lateral displacements estimated from radio frequency echo data. The derived analytical expressions include the effects of signal decorrelation, electronic noise, point spread function (PSF), and signal processing parameters (window size and overlap between the successive windows). We modeled the 2-D PSF of pulse-echo imaging system as a sinc-modulated spatial sine pulse in the axial direction and as a sinc function in the lateral direction. For validation, we compared the variance in displacements and strains, incurred when quasi-static elastography was performed using conventional linear array (CLA), plane wave (PW) and compounded plane wave (CPW) imaging techniques. We also extended the theory to assess the performance of vascular elastograms. The modified analytical expressions predicted that CLA and CPW should provide the worst and best elastographic performance, respectively, which was confirmed both in simulations and experimental studies. Additionally, our framework predicted that the peak performance should occur when 2% strain is applied, the same order of magnitude as observed in simulations (1%) and experiments (1% – 2%).

5:00

2pBA11. Numerical simulation of transcranial ultrasound imaging using a two-dimensional phased array. Petr V. Yuldashev, Sergey Tsysar, Vera Khokhlova (Phys. Faculty, M. V. Lomonosov Moscow State Univ., 119991, Russian Federation, Moscow, Leninskie Gory, Moscow 119991, Russian Federation, petr@acs366.phys.msu.ru), Victor D. Svet (N.N. Andreyev Acoust. Inst., Russian Acad. of Sci., Moscow, Russian Federation), and Oleg Sapozhnikov (Phys. Faculty, M. V. Lomonosov Moscow State Univ., Moscow, Russian Federation)

Trans-skull ultrasound imaging of brain structures is a challenging problem because of strong attenuation of the skull and reflections from its boundaries. In addition, because the speed of sound in skull is much higher than in soft tissues, nonuniform thickness and heterogeneous bone structures cause strong refraction and aberration effects. In this work, transcranial ultrasound imaging of a 3D volume was simulated numerically. A linear wave equation in an inhomogeneous medium was modeled using the k-space method. A phased array comprising 10000 identically shaped square elements distributed over 70 x 70 mm² area was used to generate a quasi-plane 2-cycle pulsed wave at 1 MHz. A spherical 3-mm diameter scatterer was placed 30 mm behind a cranial bone phantom with mass density 1900 kg/m³, sound speed 2500 m/s, and an irregular thickness varying from 5 to 8 mm. First, two reflections from the face and back sides of the phantom were used to determine its thickness. Then, a delay and sum algorithm was applied to the received echo signals to compensate for aberrations. It was shown that the scatterer was only visible when aberration compensation was applied. [Work supported by RSF-14-15-00665.]

2pBA12. Experimental implementation of a synthesized two-dimensional phased array for transcranial imaging with aberration correction. Sergey Tsysar (Phys. Dept., Lomonosov Moscow State Univ., GSP-1, 1-2 Leninskie Gory, Moscow 119991, Russian Federation, sergey@acs366.phys.msu.ru), Victor D. Svet (N.N.Andreyev Acoust. Inst., Moscow, Russian Federation), Petr V. Yuldashev, Vera Khokhlova, and Oleg Sapozhnikov (Phys. Dept., Lomonosov Moscow State Univ., Moscow, Russian Federation)

Ultrasound (US) imaging of brain structures is a challenging, but highly promising diagnostic technology in medical ultrasound. Recent advances in transcranial US therapy suggest the potential to implement diagnostic US at higher frequencies, ideally for full brain imaging. In this work, we present experimental results of ultrasound imaging of spherical and tubular scatterers placed behind a skull phantom. The phantom was produced from a casting compound with acoustic properties matching those of skull. Phantom shape was defined from CT data of a human skull and 3D printing of a mold. A two-dimensional ultrasound array was simulated by mechanical translation of the focal spot of a broadband single-element 2 MHz transducer over the phantom surface. This synthesized array mimicked a 2D flexible phased array placed on the top of the patient's head. A pulse-echo technique was used for reconstructing the thickness of the skull phantom and detecting backscattered signals from the test objects. Transcranial image reconstruction was performed using a delay-and-sum technique that accounts for refraction and absorption inside the phantom. It was demonstrated that aberration correction using either straight rays or more accurate refracted raytracing yields significant improvement of image quality. [Work supported by RSF-14-15-00665.]

2pBA13. Using frequency-sum beamforming in passive cavitation imaging. Shima Abadi (Eng. and Mathematics, Univ. of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011, abadi@uw.edu), Kevin J. Haworth (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Karla P. Mercado (Dept. of Internal Medicine, Univ. of Cincinnati, Rochester, NY), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Passive Cavitation Imaging (PCI) is a method for locating cavitation emissions to study biological effects of ultrasound on tissues. In this method, an image is formed by beamforming passively recorded acoustic emissions with an array. The image resolution depends on the ultrasound frequency and array geometry. Acoustic emissions can be scattered due to tissue inhomogeneity, which may degrade the image resolution. Emissions at higher frequencies are more susceptible to such degradation. Frequency-sum beamforming is a nonlinear technique that alters this sensitivity to scattering by manufacturing higher frequency information from lower frequency components via a quadratic product of complex signal amplitudes. This presentation evaluates the performance of frequency-sum beamforming in a scattering environment using simulations and experiments, conducted in the kHz and MHz frequency regimes. First, 50 and 100 kHz signals were broadcasted from a single source to an array of 16 hydrophones in a water tank with and without discrete scatterers. Second, a tissue-mimicking phantom perfused with microbubbles was insonified at 2 MHz, and the emissions were received by a 128-element linear array. The performance of frequency-sum beamforming was compared to conventional delay-and-sum and minimum variance beamforming in mild and strong scattering environments. [Work partially supported by NAVSEA through the NEEC.]

MONDAY AFTERNOON, 26 JUNE 2017

ROOM 205, 1:20 P.M. TO 4:40 P.M.

2pEA

Engineering Acoustics: Ducts and Mufflers II

Mats Åbom, Cochair

The Marcus Wallenberg Laboratory, KTH-The Royal Inst. of Technology, Teknikringen 8, Stockholm 10044, Sweden

David Herrin, Cochair

Department of Mechanical Engineering, University of Kentucky, 151 Ralph G. Anderson Building, Lexington, KY 40506-0503

Invited Papers

1:20

2pEA1. Hybrid silencer transmission loss above a duct's plane wave region. Paul T. Williams (AAF Ltd, Northumberland, United Kingdom), Mats Åbom (The Marcus Wallenberg Lab., KTH-The Royal Inst of Technol., Teknikringen 8, Stockholm 10044, Sweden, matsabom@kth.se), Ray Kirby (Mech. Eng., Brunel Univ., Middlesex, United Kingdom), and James Hill (AAF Ltd., Northumberland, United Kingdom)

For large ducts, the removal of low frequency and tonal noise is normally achieved through the use of inefficient dissipative silencers; however, a combination of dissipative and reactive solutions could be more effective. But reactive noise control solutions are rarely applied to large diameter duct systems since it is commonly assumed that the low cut-on frequency of higher order modes severely restricts their efficiency. However, it is possible for a reactive silencer to remain operational outside of the plane wave region, provided the reactive elements are distributed across the cross-section of the duct. Of course, at higher frequencies, the sound field within a duct will have nonplane wave modal content, and the transmission loss is expected to differ compared to the plane wave condition. This effect is investigated here using numerical (FEM) predictions for hybrid dissipative-reactive parallel baffle silencers and the performance of the reactive elements is explored under different excitations. The effects of non-planar fields and individually excited modes are analyzed, and it is found that the frequency range over which quarter wave resonators contribute to transmission loss can be extended above the cut-on frequency of the duct by increasing the number of baffles.

1:40

2pEA2. Comparison of an integral and a collocation based impedance-to-scattering matrix methods for large silencers analysis.

Peng Wang (Mech. Eng., Univ. of Kentucky, 151 Ralph G. Anderson Bldg., Lexington, KY 40506, pwa229@g.uky.edu), Limin Zhou (Akustica Inc., Pittsburgh, PA), and T. W. Wu (Mech. Eng., Univ. of Kentucky, Lexington, KY)

Large silencers used in power generation industry usually have a large cross section at the inlet/outlet. The plane-wave cutoff frequency of the inlet/outlet duct could be below only a few hundred Hz. To evaluate the acoustical performance of large silencers above the inlet/outlet cutoff, either an integral based or a point-collocation based impedance-to-scattering matrix method may be applied to convert the BEM impedance matrix to the scattering matrix with the higher-order modes at the inlet/outlet. In this presentation, these two impedance-to-scattering matrix methods are introduced first, and then several test cases are used to compare the computational accuracy, efficiency, and stability of the two methods.

2:00

2pEA3. Investigation of the effect of neglecting reflections in Power-based sound in ducts. Mina W. Nashed, Tamer Elnady (Group for Adv. Res. in Dynamic Systems (ASU-GARDS), Ain Shams Univ., 1 Elsaray St., Cairo, Abbaseya 11517, Egypt, mina.wagih@eng.asu.edu.eg), and Mats Åbom (The Marcus Wallenberg Lab., KTH-The Royal Inst of Technol., Stockholm, Sweden)

In high frequency sound propagation inside ducts, the modal density is very high that the sound starts to propagate in rays. The acoustic performance of a duct network can be simulated using power-based models. The application of such situations is for HVAC systems and large silencers for power generation. Several standards are available for the analysis of HVAC systems using the same technique, such as ASHRAE and VDI. For each element, the flow-generated sound power inside the element is added to the input sound power and the output sound power is calculated by subtracting the Insertion Loss of the element. The attenuated sound energy can be either dissipated inside the element or reflected back to the system. Standards always assume that no energy is reflected and all the attenuation happens inside the element. This assumption is investigated in this paper. Several standard HVAC elements are considered, calculating the amount of energy dissipated inside the element and that reflected back. It was found that if all the reflected energy is considered, this will affect the output power from the system, especially in the highly reflective elements. The investigation was done using the Finite Element Method with the ray tracing technique.

2:20

2pEA4. Eigenvalue analysis for acoustic multi-ports. Stefan Sack and Mats Åbom (The Marcus Wallenberger Lab., The Royal Inst. of Technol., Teknikringen 8, Stockholm 100 44, Sweden, ssack@kth.se)

Acoustic multi-ports are commonly used to describe the scattering (the transmission and reflection) and the source of aero-acoustic components in duct and pipe systems. The components are therefore modeled as “black-boxes,” assuming linear and time invariant systems. Using linear network theory, two components can be combined to a cascade for which the scattering and sources are predicted. This step, however, requires decoupled components; the flow disturbances downstream of an aero-acoustic source can be large and turbulences impinging on the downstream component may change its acoustic properties. In this presentation, we show how to use eigenvalue equations in order to investigate this so called “installation-effect” on both, the scattering and the source of induct components. The theoretical results are compared with measurements in order to conclude on the changing source and scattering mechanisms.

2:40

2pEA5. Linear sound amplification and absorption in a corrugated pipe. Xiwen Dai (CNRS, LAUM, UMR CNRS 6613, Ave. O Messiaen, F-72085 LE MANS Cedex 9, France, Le Mans 72085, France, xiwen.dai@univ-lemans.fr), Joachim Golliard (TNO, Delft, Netherlands), and Yves Aurégan (CNRS, Le Mans, France)

Linear sound propagation in an axisymmetric corrugated pipe with shear flow is studied numerically and experimentally. The acoustic and hydrodynamic perturbations are described by the linearized Euler equations (LEEs) in a parallel shear flow. Wave propagation and scattering are computed by means of a multimodal method where the disturbances are expressed as a linear combination of acoustic modes and hydrodynamic modes. The Floquet-Bloch approach is used to calculate the wavenumber in the periodic system. Both sound amplification and absorption, depending on the Strouhal number, are well predicted compared to experiments, which means that the flow-acoustic coupling in the system is effectively described by the present model. It is also shown that the corrugated pipe can amplify the sound even if the shear layer over the cavities is stable everywhere.

3:00–3:20 Break

3:20

2pEA6. Stop whistling! A note on fluid driven whistles in flow ducts. Mikael Karlsson (MWL, KTH, Teknikringen 8, Stockholm 10044, Sweden, kmk@kth.se), Magnus Knutsson (Volvo Cars, Göteborg, Sweden), and Mats Åbom (MWL, KTH, Stockholm, Sweden)

The generation mechanism and possible counter measures for fluid driven whistles in low Mach number flow duct networks are discussed. The vortex sound model, where unstable shear layers interact with the acoustic field and act as amplifiers under certain boundary conditions, is shown to capture the physics well. Further, for the system to actually whistle, an acoustic feedback to the amplifying shear layer is also needed. The demonstration example in this study is a generalized resonator configuration with annular volumes attached to a straight flow duct via a number of small holes, perforations, around the duct’s circumference. At each hole, a shear layer is formed and the acoustic reflections from the resonator volumes and the up and downstream sides provides a possible feedback to them. The attenuation properties as well as the whistling frequencies at varying inlet mean flow velocities for this system are studied both numerically and experimentally showing that good quality predictive simulations are possible using the vortex sound theory. Finally, a few countermeasures against whistling are tested. Both the feedback and the shear layers are manipulated. Best effect was found disturbing the shear layers by covering the holes with a coarse mesh.

3:40

2pEA7. A numerical investigation of Helmholtz resonators in the presence of grazing flow by means of the lattice Boltzmann method. Andre M. Spillere (Dept. of Mech. Eng., Federal Univ. of Santa Catarina, Campus Reitor Joao David Ferreira Lima, Florianopolis, Santa Catarina 88040-900, Brazil, andre.spillere@lva.ufsc.br), José P. de Santana Neto, Andrey R. da Silva (Dept. of Mech. Eng., Federal Univ. of Santa Catarina, Florianópolis, Santa Catarina, Brazil), and Julio A. Cordioli (Dept. of Mech. Eng., Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil)

Helmholtz resonators remain widely used in noise control. In applications such as aircraft engines and exhaust systems, the presence of a grazing flow significantly changes their behavior, and a correct prediction of their acoustic properties is essential to improve noise reduction. With the purpose of understanding the physical phenomena associated with the acoustic-flow interaction, the simulation of a single 2D Helmholtz resonator was considered by means of an in-house numerical code based on the lattice Boltzmann method (LBM). The results have been validated against published data based on both experimental results and direct numerical simulation (DNS) for normally incident acoustic waves in the absence of flow. The investigations will proceed by taking into account grazing acoustic waves in the presence of a grazing flow, similarly to the conditions found in a liner test rig. Efforts will be focused on typical aircraft engine inlet conditions, i.e., high Mach numbers and SPL. Both experimental and numerical results will be compared in terms of absorption coefficient and impedance.

4:00

2pEA8. Flow noise generation in a pipe bend. Magnus Knutsson, Simone Vizzini (Noise and Vib. Ctr., Volvo Car Group, Dept 91620, Göteborg 40531, Sweden, magnus.knutsson@volvocars.com), Maria Dybeck (Powertrain Eng., Volvo Car Group, Göteborg, Sweden), and Mats Åbom (The Marcus Wallenberg Lab., KTH, Stockholm, Sweden)

Noise generated by low Mach number flow in duct networks is important in many industrial applications. In the automotive industry, the two most important are the ventilation duct network and the engine exhaust system. Traditionally, design is made based on rule-of-thumb or slightly better by simple semi-empirical scaling laws for flow noise. In many cases, strong curvatures and local deviations from circular cross-sections are created due to outer geometry restrictions. This can result in local relatively high flow velocities and complex flow separation patterns and as a result, rule-of-thumb and scaling law methods can become highly inaccurate and uncertain. More advanced techniques based on time domain modelling of the fluid dynamics equations together with acoustic analogies can offer a better understanding of the local noise generation, the propagation and interaction with the rest of the system. This investigation contains a study on flow noise generation in a circular duct with a 90-degree bend carrying a low Mach number flow. Experimental results are presented and compared to numerical simulations, based on a combination of computational fluid dynamics and the acoustic analogies by Lighthill and Möhring, as well as semi-empirical models.

Contributed Paper

4:20

2pEA9. Attenuation measurements inside and at the output of a passive silencer equipped with parallel absorbing baffles. Xavier Kaiser (CEDIA, Univ. of Liege, Liege, Belgium), Sebastien Brandt (Eng. studies, Haute école de la Province de Liege, Liege, Belgium), Benoit Meys (Test cells, Safran Aero Boosters, Herstal, Belgium), Nicolas Plom (Bureau d'acoustique BANP, Liege, Belgium), and Jean-Jacques Embrechts (Elec. Eng. and Comput. Sci., Univ. of Liege, Campus du Sart-Tilman B28, Quartier Polytech 1, 10 Allée de la découverte, Liege 4000, Belgium, jjembrechts@ulg.ac.be)

The mock-up of a great passive silencer used for noise attenuation in industrial applications has been designed and tested in laboratory. This mock-up consists of three metal casing containing the noise source and

several removable rails and supports, allowing the test of different configurations of parallel absorbing baffles. The output of the silencer radiates in an anechoic chamber in order to simulate free-field conditions. The acoustic attenuation has been measured not only at the output, but also inside the silencer, with a mobile microphone located at several positions along the axis. Also, the tested configurations include three types of absorbing "cushions" and several geometrical arrangements. The results show that the maximum insertion loss (dB) measured at the output of the silencer corresponds to frequencies between 800 Hz and 2.5 kHz and its value strongly depends on the air gaps between baffles. The measurements with the mobile microphone show a linear decrease of the sound pressure level with the distance along the axis (mainly between 100 Hz and 1 kHz), which corresponds to the depth of absorbing material involved in the attenuation. Finally, a comparison with a real-scale silencer is discussed.

2p MON. PM

2pID**Interdisciplinary: Neuroimaging Techniques II**

Martin S. Lawless, Cochair

Graduate Program in Acoustics, The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Adrian KC Lee, Cochair

University of Washington, Box 357988, University of Washington, Seattle, WA 98195

Sophie Nolden, Cochair

RWTH Aachen University, Jaegerstrasse 17/19, Aachen 52066, Germany

Z. Ellen Peng, Cochair

Waisman Center, University of Wisconsin-Madison, 1500 Highland Avenue, Madison, WI 53711

G. Christopher Stecker, Cochair

*Hearing and Speech Sciences, Vanderbilt University, 1215 21st Ave. South, Room 8310, Nashville, TN 37232****Invited Papers*****1:20****2pID1. Breaking the barriers of temporal and spatial resolutions for ultrasound neuroimaging.** Mickael Tanter (Langevin Inst. (ESPCI Paris, CNRS, Inserm), Inserm, 17 rue Moreau, Paris 75012, France, mickael.tanter@gmail.com)

The introduction of plane or diverging wave transmissions rather than line by line scanning focused beams has broken the conventional barriers of ultrasound imaging. The frame rate reaches the theoretical limit of physics dictated by the ultrasound speed and an ultrasonic map can be provided typically in tens of micro-seconds (thousands of frames per second). Interestingly, this leap in frame rate is not only a technological breakthrough, but it permits the advent of completely new ultrasound imaging modes, in particular, neuro-functional ultrasound imaging of brain activity (fUltrasound). Indeed, ultrafast Doppler gives ultrasound the ability to detect very subtle blood flow in small vessels and paves the way for fUltrasound of brain activity through the neurovascular coupling. It provides the first modality for whole brain imaging on awake and freely moving animals with unprecedented resolutions^{1,2,3} compared to fMRI. Finally, we demonstrated that it can be combined with 3 μm diameter microbubbles injections to provide a first *in vivo* and non-invasive imaging modality at microscopic scales deep into organs by localizing the position of millions of microbubbles at ultrafast frame rates. This ultrasound localization microscopy technique solves for the first time the problem of *in vivo* imaging the whole brain microvasculature⁴. 1. Mace *et al.*, Nature Methods 2011. 2. Osmanski *et al.*, Nature Commun. 2014. 3. Sieu *et al.*, Nature Methods 2015. 4. Errico *et al.*, Nature 2015.

2:00**2pID2. The ins and outs of capturing brain activities associated with auditory perception and cognition.** Adrian K. C. Lee (Dept. of Speech and Hearing and Inst. for Learning and Brain Sci. (I-LABS), Univ. of Washington, Box 357988, Seattle, WA 98195, aklee@uw.edu) and G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ., Nashville, TN)

Magnetoencephalography, electroencephalography, and functional magnetic resonance imaging (MEG, EEG, and fMRI) have been used extensively to study human auditory perception and cognition. Due to the different temporal and spatial resolutions associated with each of these neuroimaging modality, each technique offers a different unique window into how our cortex participates in auditory tasks. In this talk, a number of classical paradigms will be presented and their relative strengths and shortcomings would be discussed. Other methodological advances and challenges particularly relevant to experiments in auditory perception and cognition will also be reviewed. [Work supported by NIH R01 DC013260 (AKCL) and R01DC011548 (GCS).]

2:20

2pID3. Oscillatory brain activity in response to emotional sounds in musicians and non-musicians. Sophie Nolden (RWTH Aachen Univ., Jaegerstrasse 17/19, Aachen 52066, Germany, nolden@psych.rwth-aachen.de), Simon Rigoulot (McGill Univ., Montreal, QC, Canada), Pierre Jolicoeur (Univ. of Montreal, Montreal, QC, Canada), and Jorge L. Armony (McGill Univ., Montreal, QC, Canada)

Emotions can be conveyed through a variety of channels in the auditory domain such as the human voice or music. Recent studies suggest that expertise in one sound category can impact the processing of emotional sounds in other sound categories. We focused here on how the neural processing of emotional information varies as a function of sound category and expertise of participants. Electroencephalogram (EEG) of 20 non-musicians and 17 musicians was recorded while they listened to speech prosody, vocalizations (such as screams and laughter), and musical sounds. The amplitude of EEG-oscillatory activity in the theta, alpha, beta, and gamma band was quantified and Independent Component Analysis (ICA) was used to identify underlying components of brain activity in each band. Sound category-dependent activations were found in frontal theta and alpha, as well as greater activation for musicians than for non-musicians. Differences in the beta band were mainly due to differential processing of speech. The results reflect musicians' expertise in recognition of emotion-conveying music, which seems to also generalize to emotional expressions conveyed by the human voice, in line with previous accounts of effects of expertise on musical and vocal sounds processing.

2:40

2pID4. Using functional magnetic resonance imaging to assess the emotional response to room acoustics. Martin S. Lawless and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, msl224@psu.edu)

A previous pilot study by the authors demonstrated the potential of using neuroimaging techniques to investigate a listener's emotional response to room acoustic conditions of varying preference. The hypothesis of the pilot study and the present work is that regions associated with reward and pleasure will activate when an individual listens to pleasing room acoustics contrasted with listening to unpleasant room acoustics. In this study, auralizations were generated in simulated room conditions ranging from anechoic to extremely reverberant with the expectation that the most-liked stimuli would have reverberation times between 1.0 and 2.8 s. Participants were screened based on their ability to discern differences in preference across the stimuli. Following the screening, eligible participants rated the stimuli according to overall preference in a mock MRI machine. The results from the mock MRI testing were used to identify each participant's most-liked and most-disliked stimuli, and to familiarize the participants with the MRI environment. In a second session, this pair of stimuli, along with anechoic and scrambled-music stimuli were presented to the subjects in an MRI machine. Contrasts between these conditions were analyzed to investigate if activations were present in regions associated with reward processing, including the nucleus accumbens, caudate nucleus, and orbitofrontal cortex.

Contributed Paper

3:00

2pID5. Reduced vessel tone leads to vasodilation and decreased cerebral rigidity. Katharina Schregel, Miklos Palotai (Radiology, Brigham and Women's Hospital, 221 Longwood Ave., Boston, MA 02115, kschregel@bwh.harvard.edu), Navid Nazari (Biomedical Eng., Boston Univ., Boston, MA), Paul E. Barbone (Mech. Eng., Boston Univ., Boston, MA), Ralph Sinkus (Biomedical Eng., Kings College London, London, United Kingdom), and Samuel Patz (Radiology, Brigham and Women's Hospital, Boston, MA)

Magnetic Resonance Elastography (MRE) measures elastic shear wave propagation *in vivo* to infer biomechanical properties non-invasively [Muthupillai, *et al.* (1995) *Science*;269:1854-1857.]. Models predict that tissue stiffness is influenced by changes of vascular properties [Parker, *et al.*

(2016) *PhysMedBiol*;61:4890-4903.]. Cerebral blood supply is closely regulated by diameter changes of blood vessels. We here investigated the influence of vasodilation on cerebral brain stiffness with MRE. A healthy C57BL/6 mouse was anesthetized with isoflurane mixed in 100% O₂. Vasodilation was induced by a hypercapnic challenge (isoflurane mixed in 95% O₂ + 5% CO₂). Brain stiffness was measured with a 3D spin-echo MRE sequence in a 7T animal MRI scanner under normocapnic and hypercapnic conditions. Vibration frequency was 1 kHz. Wave length and wave speed was observed to decrease significantly under hypercapnic conditions across the whole brain when compared to baseline. These changes correspond to a decrease in tissue rigidity. We conclude, therefore, that vasodilation via reducing vessel tone leads to a significant decrease in brain rigidity. Therefore, potential changes in cerebral blood flow due to physiological or pathological conditions should be considered when studying brain rigidity.

2p MON. PM

2pMU

Musical Acoustics: Electronically-Augmented Instruments

Edgar J. Berdahl, Cochair

Music, Louisiana State University, 102 New Music Building, Baton Rouge, LA 70803

Adrien Mamou-Mani, Cochair

*IRCAM, 1 place Stravinsky, Paris 75004, France***Chair's Introduction—1:15***Invited Papers***1:20****2pMU1. Trekking around ancestors of smart instruments.** Charles Besnainou (Lutheries - Acoustique - Musique, Université Pierre et Marie Curie, chez Baudry, Saint Eugène 17520, France, charles.besnainou@upmc.fr)

In the present time, smart instruments are the ongoing revolution tying together acoustical instruments and computers. The concepts of electronic active control of structure are the heart of the process. The aim of this paper is to summarize more than one century of trials. Who remembers the “infinite sound” piano of Richard Eisenmann driven by electromagnetics and been touched, during the *Industrial Art Exhibition* in Munich in 1888, by Hermann von Helmholtz, and earlier the Pape’s enhanced piano with air blow? Some experiments revive the project as the E-bow dedicated for the steel-string guitar. Then, after a jump in the 1990’s as exercises for students on teaching feedback Proportional-Integral-Derivative analogical central heating regulator. On the cliff of the positive feedback applied for a xylophone bar, and so on just before numerical systems.

1:40**2pMU2. Bela: An embedded platform for low-latency feedback control of sound.** Andrew McPherson (School of Electron. Eng. and Comput. Sci., Ctr. for Digital Music, Queen Mary Univ. of London, Mile End Rd., London E1 4NS, United Kingdom, andrewmcphersonasa@gmail.com)

Bela is an open-source embedded platform for audio and sensor processing. It uses a Beagle Bone Black single-board computer with a custom hard-real time audio environment based on Xenomai Linux, which is capable of submillisecond round-trip audio latencies (as low as 80 microseconds in certain configurations). The Bela hardware features stereo audio input and output, 8 channels each of 16-bit analog input and output, and 16 digital I/Os, all sampled at audio rates with nearly jitter-free alignment to the audio clock. This paper will present the hardware, software, and selected applications of Bela. Bela is suitable for creating digital musical instruments and interactive audio systems, and its low latency makes it especially well adapted for real-time feedback control over acoustic systems. It has been used in feedback control experiments with wind and string instruments and used as the basis for a study of the performer’s experience of latency on percussion instruments.

2:00**2pMU3. Modal active control: A tool to finely adjust the sound of string instruments.** Simon Benacchio (IRSST, 505 Boulevard de Maisonneuve O, Montréal, QC H3A 3C2, Canada, Simon.Benacchio@irsst.qc.ca)

Controlling the vibratory properties of musical instruments is an important challenge for musical acousticians, musicians, and instrument makers. The latter try to control these properties modifying mechanical parameters of instruments like their shape or their materials to obtain some expected sound attributes. Musicians also modify the vibratory properties of their instruments to change their sound using mutes, for example. Musical acousticians try to modify these properties because it is an intuitive way to investigate the relationship between the instrument mechanisms and their sound attributes. Inspired by industrial techniques, active control was revealed as a convenient way to answer to the latter goal. Moreover, modal active control is a preferred method for the application to musical instruments since their modal parameters are believed to be good descriptors of their vibratory properties. This study aims at applying modal active control on string instruments. First, the possibilities offered by this technique are presented and experimented on several instruments. Then, examples of the control of specific phenomena are given. Couplings between the soundboard and strings for both a cello and a guitar are controlled to cancel the well-known wolf-note phenomenon for the former and to switch from strong to weak coupling for the latter.

2:20

2pMU4. Development of a hybrid wind instrument—Some key findings. Kuriijn Buys, David Sharp, and Robin Laney (Faculty of Mathematics, Computing and Technol., Walton Hall, Milton Keynes, Buckinghamshire MK7 6AA, United Kingdom, kuriijn.buys@open.ac.uk)

A hybrid wind instrument is constructed by putting a theoretical excitation model (such as a real-time computed physical model of a clarinet embouchure) in interaction with a real wind instrument resonator. In previous work, the successful construction of a hybrid wind instrument has been demonstrated, with the interaction facilitated by a loudspeaker and a microphone placed at the entrance of a clarinet-like tube. The present paper focuses on some key findings, concentrating particularly on the “musical instrument” and “research tool” perspectives. The limitations of the hybrid set-up are considered. In particular, the choice of the loudspeaker used in the set-up is explained and the occurrence (and prevention) of instabilities during the operation of the hybrid instrument are discussed. For the design of excitation models used to drive the hybrid instrument, the usefulness of dimensionless and reduced parameter forms is outlined. In contrast to previously reported physically based excitation models, it is demonstrated that a purely mathematical “polynomial model” enables an independent control of separate sound features. For all excitation models, the sounds produced with the hybrid instrument are shown to match to those predicted by simulation. However, the hybrid instrument is more easily destabilized for certain extreme parameter states.

2:40

2pMU5. Traveling wave control of stringed musical instruments. Liam Donovan and Andrew McPherson (Queen Mary Univ. of London, Queen Mary Univ., Mile End Rd., London E1 4NS, United Kingdom, l.b.donovan@qmul.ac.uk)

Traveling wave control is a technique in which the energy propagating around a structure in the form of waves can be manipulated directly in order to change the overall dynamic behaviour of the structure. In this research, traveling wave control is applied to a musical instrument string with a view to affecting the timbre of the sounds produced by the string when vibrating. A highly linear custom optical sensor is built which is capable of detecting a wave traveling on a string in a single direction, and a piezo stack actuator is situated under the termination point of the string allowing the reflection of the wave to be manipulated directly. Various controllers are analyzed theoretically in terms of their performance, stability, and musical usefulness. They are then implemented and evaluated in terms of their relevance to the design of new musical instruments.

3:00

2pMU6. Astounding sounds, amazing music—At the crossroads of audio control. Joseph A. Paradiso (Media Lab, MIT, MIT Media Lab, E14-548P, Cambridge, MA 02139, joep@media.mit.edu)

The ways in which we produce, compose, discover, and consume music have changed profoundly in only the last decades as the conveyances of these capabilities have digitally converged. What does it mean to “play” a musical instrument and what will a musical performance become? In this presentation, I will explore these fringes via recent projects from my research team at the MIT Media Lab. This includes frameworks to enable composers to exploit sources of “big” data to realize their music (ranging from physics detectors at the Large Hadron Collider to a sensor-laden former cranberry bog turning into a wetland). In a related vein, I will introduce an interactive sonification framework we have devised to dynamically rotate complex data from visual to audio, with the goal of optimally engaging eyes and ears. At the other extreme, I will describe a set of new physical instruments for musical control, such as stretchable fabric keyboards, instruments designed to be breakable during performance, and collaborative instruments that leverage social media and the Internet of Things. Finally, I will give my perspective on the recent resurgence of modular synthesizers, grounded in having built and designed perhaps the world’s largest homemade modular system between 30 and 40 years ago.

3:20–3:40 Break

Contributed Papers

3:40

2pMU7. Monitoring saxophone reed vibrations using a piezoelectric sensor. Alex Hofmann, Vasileios Chatziioannou, Alexander Mayer (Music Acoust. (IWK), Univ. of Music and Performance Art Vienna, Anton von Webern Platz 1, Vienna, Select State 1030, Austria, hofmann-alex@mdw.ac.at), and Harry Hartmann (Fiberreed, Leinfelden-Echterdingen, Germany)

In sound production on single-reed woodwind instruments the reed is oscillating in a frequency related to the length of the resonator. Strain gauge sensors attached to single reeds have been used to capture the vibrations of the reed to investigate articulation techniques on saxophone and clarinet. Reeds can be made from natural cane and also from synthetic materials like oriented polymers or layers of fiber-reinforced polymers. Such synthetic reeds allow to integrate sensors inside the reed during manufacture. However, integrated strain gauge sensors produced signals with high noise, which have been shown not to be ideal for amplification purposes. Replacing the integrated strain gauge with a piezo film sensor greatly enhanced the sound quality of the sensor reeds. With this procedure, electronically augmented woodwind instruments may be constructed for performance, acoustic measurements, and music pedagogy feedback systems.

4:00

2pMU8. Active control of Chinese gongs. Marguerite Jossic (Institut d’Alembert, UPMC CNRS UMR 7190, 4 Pl. Jussieu 75005, Institut d’Alembert, Tour 55-65, 4, Pl. Jussieu, Paris Cedex 05 75252, France, marguerite.jossic@upmc.fr), Vivien Denis, Olivier Thomas (Arts et Métiers ParisTech, LSIS UMR CNRS 7296, 8 bd. Louis XIV, Lille, France), Adrien Mamou-Mani (IRCAM CNRS UPMC UMR 9912, 1 pl. Stravinsky 75004, Paris, France), Baptiste Chomette (Institut d’Alembert, UPMC CNRS UMR 7190, 4 Pl. Jussieu 75005, Paris, France), and David Roze (IRCAM CNRS UPMC UMR 9912, 1 pl. Stravinsky 75004, Paris, France)

Active control provides one of the most promising ways of modifying instruments’ sound. However, among the various control techniques covered by this discipline, most of the experimental applications have so far been limited to instruments keeping a linear behavior in normal playing conditions. This study explores the possibility of applying active control to Chinese gongs. These instruments exhibit geometric nonlinearities in their dynamical behavior such as a very characteristic pitch glide of the fundamental mode. The implementation of a nonlinear control of this pitch glide is introduced following a two-steps process. First, a reduced nonlinear model of the instrument

dynamics is determined from the von Karman and normal form theories. This formulation allows for describing the nonlinear dynamics of the fundamental mode by one single nonlinear mode governed by a Duffing equation. The experimental identification of the Duffing model parameters is performed by measuring the backbone curve of the fundamental mode in forced vibrations. Second, the methodology for the nonlinear control is developed. In particular, the determination of the control law and the overall stability of the control system are discussed. Control simulations results as well as perspectives for experimental applications are finally exposed.

4:20

2pMU9. No latency feedback controller coupled to real-time physical models using programmable hardware. Florian Pfeifle (Univ. of Hamburg, Neue Rabenstrasse 13, Hamburg 20354, Germany, Florian.Pfeifle@uni-hamburg.de)

Over the last years, advances in technology and methodology made it possible to simulate and synthesize highly realistic finite difference (FD) models in real-time or close to real-time. Still, most conventional processing platforms introduce latency to the signal processing chain due to sequential processing and/or communication protocol timing and throughput restrictions. This can act as a severe penalty when developing expressive controller interfaces for large geometry physical models. Using field programmable gate array (FPGA) hardware enables highly customized interface and FD model designs that are able to meet *hard* real-time requirements even for large physical models. In this work, a modified five-string *banjo* coupled to a real-time physical modeling synthesis application running on an FPGA development board is presented. The proposed methodology is an extension of an existing PCIe enabled interface which was primarily developed for research applications. The new interface is aimed at facilitating expressive

interaction for musical purposes. The bi-directional excitation and feedback is realized using standard electro-mechanical sensors and actuators, making it possible to connect the string vibration of the modified banjo to arbitrary geometries as well as capturing the response of the modeled structure and feeding it back to the acoustic instrument. FD-models of different shapes and materials are implemented resulting in physically impossible instrument configurations yielding unique tone production capabilities.

4:40

2pMU10. Audio-haptic interaction with modal synthesis models. Edgar J. Berdahl (Music, Louisiana State Univ., 102 New Music Bldg., Louisiana State University, Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu)

Using a musical instrument that is augmented with haptic feedback, a performer can be enabled to haptically interact with a modal synthesis-based sound synthesizer. This subject is explored using the **resonators** object in Synth-A-Modeler. For synthesizing a single mode of vibration, the **resonators** object should be configured with a single frequency in Hertz, a single T_{60} exponential decay time in seconds, and a single equivalent mass in kg. Changing the mass not only changes the level of the output sound, it also changes how the mode feels when touched using haptic feedback. The **resonators** object can further be configured to represent a driving-point admittance corresponding to arbitrarily many modes of vibration. In this case, each mode of vibration is specified by its own frequency, decay time, and equivalent mass. Since the modal parameters can be determined using an automated procedure, it is possible to (within limits) approximately calibrate modal models using recordings of sounds that decay approximately exponentially. Various model structures incorporating the **resonators** object are presented in a variety of contexts. The musical application of these models is demonstrated alongside presentation of compositions that use them.

MONDAY AFTERNOON, 26 JUNE 2017

ROOM 203, 1:15 P.M. TO 5:40 P.M.

2pNSa

Noise, Architectural Acoustics and ASA Committee on Standards: Noise Impacts and Soundscapes on Outdoor Gathering Spaces II

Brigitte Schulte-Fortkamp, Cochair

Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 101789, Germany

K. Anthony Hoover, Cochair

McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362

Chair's Introduction—1:15

Invited Papers

1:20

2pNSa1. Soundscape design of an open-air concert venue using virtual reality technologies. Andy Chung (Smart City Maker, Hong Kong Plaza, Hong Kong HKSAR, Hong Kong, ac@smartcitymaker.com), W. M. To (Macao Polytechnic Inst., Macao, Macao), and Brigitte Schulte-Fortkamp (TU Berlin, Berlin, Germany)

Hong Kong has two mega-projects underway that include open-air concert venues. One of them is called the Kai Tak Sports Park and another is the West Kowloon Cultural District. As an open-air concert venue will produce amplified music and noise from audience, as well as vehicular traffic noise and crowd noise before and after the concerts, *a priori* analysis of soundscape and noisescapes in the district due to musical events is highly desirable. This paper reviews some noise incidents due to concerts at the Hong Kong Stadium and Chinese operas at other public places. We suggest that virtual reality technologies shall be utilized so that stakeholders can hear the possible soundscape and noisescapes at different periods of a musical event during the public engagement.

2pNSa2. How does activity affect soundscape assessments? Insights from an urban soundscape intervention with music. Daniel Steele, Cynthia Tarlao (MIL & CIRMMT, McGill Univ., 3661 Peel St, Montreal, QC H3A 1X1, Canada, daniel.steele@mail.mcgill.ca), Edda Bild (Universiteit van Amsterdam, Amsterdam, Netherlands), Julian Rice, and Catherine Guastavino (MIL & CIRMMT, McGill Univ., Montreal, QC, Canada)

The relationship between activity and soundscape has recently garnered research attention, particularly in public spaces. In the summer of 2015, we installed an interactive sound system (Musikiosk) in a busy public park allowing users to play their own content over high-quality speakers. Questionnaires (N = 197) were administered over 3 conditions: pre-installation with park users, during the installation phase with Musikiosk users, and during the installation phase with park users not using Musikiosk. For users and observers of Musikiosk, a separate evaluation of the Musikiosk intervention was also included. The questionnaire included quantitative evaluations (soundscapes scale from Swedish Soundscape Quality Protocol, restorativeness, mood, noise sensitivity), free response data (soundscape description, self-reported activity, sound source identification, reasons for park visit), and demographic info (age, interaction with others, proximity of residence). The qualitative descriptions of activity and sound sources were categorized into emergent themes. Presented here is the analysis of the interaction between activity and soundscape assessment in terms of quantitative variables and qualitative descriptions.

Contributed Papers

2:00

2pNSa3. A perception-based protocol for the systematic selection of urban sites with specific soundscapes. Bert De Coensel and Dick Botteldooren (Information Technol., Ghent Univ., iGent, Technologiepark-Zwijnaarde 15, Ghent B-9052, Belgium, bert.decoensel@ugent.be)

The Urban Soundscapes of the World project aims to set the stage for a standard on recording and reproducing urban acoustic environments with soundscape in mind. Immersive audiovisual recordings, which combine high quality spatial (binaural) audio with 360 degree video, are valuable to serve as an ecologically valid baseline for assessing the perceptual influence of noise control and soundscaping measures through auralization. As architects and designers commonly work by example, one of the goals of this project is to compile a comprehensive reference database of well-documented exemplars. These are to be recorded at a range of urban sites with a wide variety of soundscapes, in order to be able to achieve a good statistical power in any subsequent analysis. For this purpose, a protocol for selecting recording locations and time periods in a systematic way is developed, based on a common questionnaire that is conducted among panels of local experts in each selected city. The questionnaire contains open questions to look for public spaces inside the city that are perceived in various ways, regarding the presence of sound sources, the perceived affective quality and the appropriateness of the sound environment.

2:20

2pNSa4. A trial investigation to understand the characteristics of soundscape in a busy town from the viewpoint of sound quality. Takeshi Akita (Dept. of Sci. and Technol. for Future Life, Tokyo Denki Univ., 5 Senju-Asahi-cho Adachi-ku, Tokyo 1208551, Japan, akita@cck.dendai.ac.jp)

To reveal the characteristics of soundscape of a town that has an open and busy place like a shopping avenue, an investigation that requests subjects finding out attracted sounds and evaluating their sound quality when they walk around the town is tried. Eight subjects participated in the present investigation. It was carried out at Kita-Senju town in Tokyo. Each subject was required to walk around assigned area that has busy shopping avenue and road, to write down the name of sound, to take the photograph of sound source, and to evaluate its sound quality when his attention was attracted by some sounds. Sound quality was evaluated by 5 step scales that inquired strength, fineness, and acuteness. Results show that there are many kinds of sound that has little strength impression except road traffic noise. On the other hand, the evaluated results for fineness and acuteness are different among sounds that are evaluated not so loud. They also show that there are many meaningless sounds and machine oriented noises. It is suggested that finding out and evaluation method for attracted sounds in busy areas make the characteristics of soundscape clear and it will contribute to create fine sonic environment.

2:40

2pNSa5. Measuring sounds with a grid method with examining public spaces. Yalcin Yildirim (Urban Planning and Public Policy, Univ. of Texas at Arlington, 601 W Nedderman Dr. #203, Arlington, TX 76019, yalcin.yildirim@mavs.uta.edu)

This study provides an outlook of the association between sound and public space by performing sound level meter in Dallas Fort-Worth metropolitan area. The main research question of the study is that whether the characteristics (program elements, position of public spaces and roads, and public space usage in different time intervals) of the public open spaces have relationship with sound levels. In order to demonstrate it, the applied sound level meter instruments for sound environment in public open spaces by using grid method. This study recommends that time intervals have effects on sound and practice should concentrate for this essential element. At this point, sound is forgotten component and it is not paid attention by urban planners, architects, landscape architects, civil engineering and many other disciplines. Presently, there are a few studies that are related to sound relationships in the United States. Hence, this research illustrates a point of view of a sound research for rapid growing urban areas.

3:00

2pNSa6. Soundscape of Arctic settlements: Longyearbyen and Pyramiden. Dorota Czopek, Pawel Malecki, Janusz Piechowicz, and Jerzy Wiciak (AGH Univ. of Sci. and Technol., al. Mickiewicza 30, Kraków 30-059, Poland, dorota.czopek@agh.edu.pl)

This paper presents the soundscape analysis of two settlements in Spitsbergen in Svalbard archipelago. The first one is the largest settlement in Spitsbergen, Longyearbyen, with population of about 2000 people. It is the administrative center of Svalbard with airport and the seat of Governor of Svalbard. The second one, Pyramiden, is Russian, coal-mining settlement closed in 1998. Since 2007 Pyramiden has become the tourist attraction with hotel and small museum. Only a few workers live there permanently. Two one-week research expeditions were organized to perform preliminary Arctic soundscape measurements. First, summer expedition during polar day and second winter-spring expedition during the transition period between the polar night and polar day. Long and short term sound pressure level measurements together with the ambisonic recordings of unique and typical sounds were made. Both qualitative assessment and quantitative analysis of the results were carried out. The identification and classification of the existing sound sources were conducted. Furthermore, noise maps of both places together with the comparative analysis were performed.

3:20–3:40 Break

2pNSa7. Introduction and management of noise low emission zones: LIFE MONZA project. Raffaella Bellomini (Universita' Di Firenze, Via Stradivari 19, Firenze 50127, Italy, raffaella.bellomini@vienrose.it), Rosalba Silvaggio (ISPRA, Rome, Italy), Sergio Luzzi (VIE EN.RO.SE. Ingegneria, Firenze, Italy), and Francesco Borchi (Universita' Di Firenze, Firenze, Italy)

The introduction of *Low Emission Zones*, an urban area subject to road traffic restrictions in order to ensure compliance with the air pollutants limit values, set by the European Directive on ambient air quality (2008/50/EC), are common and well-established actions in the administrative government of the cities and the impacts on air quality improvement are widely analyzed, while the effects and benefits concerning the noise have not been addressed in a comprehensive manner. The definition, the criteria for analysis, and the management methods of a *Noise Low Emission Zone* are not yet clearly expressed and shared. LIFE MONZA project (Methodologies for Noise low emission Zones introduction and management—LIFE15 ENV/IT/000586) addresses these issues. The first objective of the project, cofunded by the European Commission, is to introduce an easy-replicable method for the identification and the management of the *Noise Low Emission Zones*, an urban area subject to traffic restrictions, whose impacts and benefits regarding noise issues will be analyzed and tested in the pilot area of the city of Monza, located in North Italy. Background conditions, structure, and objectives of the project will be discussed in this paper.

4:00

2pNSa8. Beyond the Noise: Open Source Soundscapes. A mixed methodology to analyze and plan small, quiet areas on the local scale, applying the soundscape approach, the citizen science paradigm, and open source technology. Antonella Radicchi (Institut für Stadt- und Regionalplanung, Technische Universität Berlin, Hardenbergstraße 40 a, Sekr. B 4, Berlin, Berlin 10623, Germany, antonella.radicchi@tu-berlin.de)

Today, cities have become increasingly noisier. In Europe, over 125 million people are affected by noise pollution from traffic every year, and apparently, quietness is becoming a luxury available only for the *elites*. There is a growing interest in protecting and planning quiet areas, which has been recognized as a valid tool to reduce noise pollution. However, developing a common methodology to define and plan quiet areas in cities is still challenging. The “Beyond the Noise: Open Source Soundscapes” project aims to fill this gap of knowledge by applying the soundscape approach, the citizen science paradigm and open source technology, with the ultimate goal of making quietness as a commons. Accordingly, a new mixed methodology to analyze and plan small, quiet areas on the local scale has been tested through the development of a pilot study in a Berlin neighborhood affected by environmental injustice and noise pollution. In this pilot study, a number of citizens have been involved in crowdsourcing data related to “everyday quiet areas” by using novel mobile technologies. This contribution illustrates the project’s theoretical background, the methods applied, the first findings of the study and its potential impact.

4:20

2pNSa9. Assessment of the relation between psychoacoustic parameters and the subjective perception of urban soundscapes. Daniel de la Prida, Antonio Pedrero, César Díaz, and María Ángeles Navacerrada (Grupo de Investigación en Acústica Arquitectónica, Tech. Univ. of Madrid, UPM: Escuela Técnica Superior de Arquitectura, Avenida Juan de Herrera n° 4, Madrid, Madrid 28040, Spain, d.delaprida@alumnos.upm.es)

Since soundscapes are strongly related to the human perception, the sound pressure level does not seem to be a sufficient representative of a soundscape by itself. Therefore, the characterization of a soundscape might be improved by using psychoacoustic parameters. A study has been conducted and the relationship between psychoacoustic parameters and the

subjective perception has been analyzed, for a collection of urban spaces. For that purpose, several locations at the city of Madrid have been selected, based on their main use and geometrical features. Then, binaural recordings have been made for several days and different seasons, which allow to observe the relevance of the behavioral differences for the same locations under different conditions. Psychoacoustic parameters, as well as sound pressure level, have been calculated for both the complete recordings and selected parts of them. A semantic differential listening test have been carried out to look for correlations between the calculated parameters and the subjective perception of a panel of participants. Finally, an automatic clustering is presented for the collection of locations. The adequacy of the proposed clustering method is evaluated by comparing the clusters to the psychoacoustic parameters and the subjective responses of the listening test.

4:40

2pNSa10. Music to some, noise to others; reducing outdoor music festivals’ sonic impact on surrounding communities. Case study: KAABOO 2016. Pantelis Vassilakis (AcousticsLab - Acoust. Consulting, 616 W Imperial Ave., #4, El Segundo, CA 90245, pantelis@acousticlab.org) and Aaron Davis (Audio, ECTO Productions Inc., Bensenville, IL)

For music scenes to coexist and thrive alongside residential communities, approaches to the problem of music as noise must acknowledge the impact of noise signal type on listener annoyance levels. The challenge has yet to be properly addressed by the environmental acoustics community, which focuses on measurement standards and mitigation techniques applicable to mechanical noise but unfit to address noise issues related to music. Differences include short versus long range contexts, health versus annoyance considerations, and continuous/unintelligible versus time-variant/intelligible source signals. Noise ordinances often introduce further complications, requiring disambiguation to provide valid/assessable expectations. The presentation outlines how the problem was successfully tackled for KAABOO 2016, a large-scale open-air music festival involving over 100 acts, over 75,000 patrons, and multiple outdoor stages. We discuss (a) working with the city and venue to fine-tune noise ordinance expectations and support valid compliance assessment; (b) designing and deploying sophisticated sound systems, powerful enough to fulfill audience expectations and focused enough to effectively reduce noise impact on the surrounding communities; (c) cooperating with the artists’ teams to appropriately reduce on-site levels; and (d) obtaining relevant noise data prior and during the event to validly capture the event’s noise impact and formally assess compliance.

5:00

2pNSa11. Soundscapes, social media, and big data: The next step in strategic noise mapping. Eoin A. King, She’ifa Punla-Green, and Samuel Genovese (Acoust. Program and Lab, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, eoking@hartford.edu)

The current state-of-the art in noise assessment involves the development of a strategic noise map to identify areas with excessive noise levels, expressed in terms of a single time-averaged noise indicator. While noise maps yield important information regarding sound pressure levels in a particular space, they do not give any representation of the overall sound quality in that space. A more human-centered approach to noise assessment could be achieved by developing soundscapes as a complementary tool to noise mapping. However, most soundscape studies traditionally use surveys or interviews to assess general sentiment toward the acoustic environment, and as such are generally restricted to small geographic areas, compared to the entire cities considered in noise mapping. Instead of using traditional assessment techniques, this project aims to harness the potential of big data, including, for example noise complaint data or social media chatter related to noise, to better assess public sentiments towards soundscapes. This would yield an unparalleled dataset of public opinions and perceptions of the acoustic environment. Initial results based on an analysis of NYC311 complaints and geo-localized data mined from Twitter are presented.

5:20–5:40 Panel Discussion

2pNSb**Noise, Physical Acoustics, ASA Committee on Standards, and Structural Acoustics and Vibration:
Sonic Boom Noise III: Community Exposure and Metrics**

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

Invited Papers

1:20

2pNSb1. NASA's Low Boom Flight Demonstration: Assessing community response to supersonic overflight of quiet supersonic aircraft. Peter Coen, Alexandra Loubeau (NASA, NASA Langley Res. Ctr., MS 264, Hampton, VA 23681, peter.g.coen@nasa.gov), and Brett Pauer (NASA, Edwards, CA)

Innovation in Commercial Supersonic Technology is one of six thrusts that guide NASA's Aeronautics Research. The near term objective of this activity is establishment of standards for acceptable overland supersonic flight, in cooperation with international standards organizations. To accomplish this objective, NASA believes the next step is to conduct a flight demonstration using a research aircraft designed to produce not a sonic boom, but a quieter "thump" sound. Based on the success of recent research, NASA has initiated design studies on a Quiet Supersonic Technology (QueSST) Aircraft. The Flight Demonstration will culminate in a series of campaigns in which the QueSST aircraft will be flown over communities. Surveys will be conducted to develop a database of public response to the sounds. This data will support ongoing international efforts to develop the noise certification standard. While the first flight of the aircraft is still a few years away, NASA recognizes there is much to be done to prepare for the community response test phase. This includes community identification and engagement, survey and instrumentation design and local and Federal government approvals. The paper will present background and of NASA planning to date, and solicit input from the research community on next steps.

1:40

2pNSb2. An examination of the variations in estimated models for predicting annoyance due to supersonic aircraft noise. Daniel J. Carr and Patricia Davies (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 177 South Russell St., West Lafayette, IN 47907-2099, daviesp@purdue.edu)

There is a need for good criteria to evaluate the acoustic outcomes of designs of future commercial supersonic aircraft. Such criteria could be used with sound predictions to assess impact on communities under flight paths of supersonic aircraft. While surveys of communities exposed to supersonic aircraft noise should be part of the criteria development and validation, some candidate models of people's responses need to be developed to help focus the design of the community tests. While several tests have been conducted, the models proposed to predict annoyance differ. Analysis of response data from several sonic boom subjective tests is presented. Either indoor or outdoor sounds have been used in the tests, and the models are based on metrics from indoor and from outdoor sounds. The effects of the environment in which the people hear the sounds, the signal sets used in the tests and in the analysis, the metrics and types of models used are discussed.

2:00

2pNSb3. Dose-response model comparison of recent sonic boom community annoyance data. Jonathan Rathsam (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, jonathan.rathsam@nasa.gov) and Laure-Anne Gille (Shanghai, China)

To enable quiet supersonic passenger flight overland, NASA is providing national and international noise regulators with a low-noise sonic boom database. The database will consist of dose-response curves, which quantify the relationship between low-noise sonic boom exposure and community annoyance. The recently-updated international standard for environmental noise assessment, ISO 1996-1:2016, references two fitting methods for dose-response analysis. Fidell's community tolerance method is based on theoretical assumptions that fix the slope of the curve, allowing only the intercept to vary. By contrast, Miedema and Oudshoorn's method is based on multilevel grouped regression. These fitting methods are applied to an existing pilot sonic boom community annoyance data set from 2011 with a small sample size. The purpose of this exercise is to develop data collection and analysis recommendations for future sonic boom community annoyance surveys.

2:20

2pNSb4. A multiple-criteria decision analysis to evaluate sonic boom noise metrics. Joe DeGolia (Elec. Eng., Univ. at Buffalo, 12 Capen Hall, Buffalo, NY 14260, jedegoli@buffalo.edu) and Alexandra Loubeau (Structural Acoust., NASA Langley Res. Ctr., Hampton, VA)

A statistical analysis was performed to select which noise metrics are best at explaining human annoyance to sonic boom noise. This follows previous work to downselect these metrics, but offers a more robust argument. The analysis began with calculation of a set of thirteen noise metrics and collected information about their explanatory powers (r^2 between annoyance rating and noise metric) in five laboratory dose-response studies performed at different sonic boom simulation facilities. In these studies, indoor and outdoor human responses were gathered under various experimental conditions—booms alone, booms with rattle indoors, and booms with indoor vibration. This input data was then passed through two stages of multiple-criteria decision-making algorithms. In the first step, a Pareto efficiency analysis was conducted to objectively group metrics and eliminate poor contenders. The second step involved an Analytic Hierarchy Process to rank the remaining metrics, with an option to subjectively weight the studies by perceived importance. The result of this downselection is the ranking of five metrics that were selected as top contenders: BSEL, ESEL, ISBAP, PL, and DSEL. [ISBAP = PL + 0.4201(CSEL-ASEL)].

2:40

2pNSb5. Some practical difficulties in assessing community response to low-amplitude sonic booms. Sanford Fidell (Fidell Assoc., Inc., 23139 Erwin St, Woodland Hills, CA 91367, sf@fidellassociates.com), Richard Horonjeff (Consultant in Acoust. and Noise Control, Boxborough, MA), and Vincent Mestre (Landrum and Brown, Irvine, CA)

NASA Langley Research Center has been engaged for several years in planning tests of public acceptance of exposure to low-amplitude sonic booms that will be created by its Quiet Supersonic Transport (QueSST) X-plane design. Estimation of a dosage-response relationship for the prevalence of high annoyance with sonic booms is a key part of this testing. The need to credibly assess prompt, single-event responses within carpet boom corridors extending along hundreds of miles of flight tracks and their linkage to sonic boom sound levels at respondents' homes is a large part of the challenge of establishing such a relationship. As many as tens of thousands of contact attempts and thousands of completed interviews must be achieved within short time periods. Conventional social survey approaches to measuring cumulative noise exposure and the prevalence of high annoyance in airport environs are ill-suited to such purposes. ADS-B-based, Internet-enabled flight tracking and impulse noise measurement, as well as high speed interviewing methods, are currently under investigation as potential solutions to difficulties in synchronizing interviews with arrival times of shock waves at residences of exposed populations.

3:00

2pNSb6. Development of a metric utilizing outdoor sonic boom for correlating indoor human annoyance responses. John M. Morgenstern (Lockheed Martin Aeronautics, 1011 Lockheed Way, Palmdale, CA 93599, john.morgenstern@lmco.com)

In 2015, the CAEP (Committee for Aviation Environmental Protection) SSTG (SuperSonic Task Group) assessed over 70 metrics for resolving acceptability of sonic booms. A ground rule was metric application to the outdoor sonic boom signature. Studies indicate people spend 90% of time indoors and sonic boom is expected to be the same or more annoying indoors. NASA developed an IER (Indoor Environment Room) facility to simulate indoor sonic booms and collected 30 humans' responses to 140 representative booms. Because all other metrics were based on human perception of loudness without indoor effects, a new metric was developed. Previous work suggested indoor annoyance from sonic booms was predominantly based upon indoor loudness, building response and associated rattle. This new metric combines one metric for indoor loudness and one metric for building response and rattle. Indoor loudness [PL(i)] is based upon the highest correlated outdoor metric, PL, but calculated after adjusting 1/3 octave levels for transmission loss. Building response (BR) averages of 1/3 octave levels in the 10-40 Hz range. The combined metric correlated the NASA indoor data with an R^2 of 0.939, higher than the next best metrics: ISBAP 0.921, PL(i) 0.910 and PL 0.881.

3:20–3:40 Break

3:40

2pNSb7. Effects of model fidelity on indoor sonic boom exposure estimates. Jacob Klos (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., MS 463, Hampton, VA 23681, j.klos@nasa.gov)

Commercial supersonic flight is prohibited over land, but this may change in the near future with the introduction of supersonic aircraft that produce a substantially quieter sonic boom. A transient modal interaction model is used to simulate the acoustic and vibration environment inside a large ensemble of residential homes to estimate the range in levels to which residents may be exposed during overflight of low-boom supersonic aircraft. However, the choice of fidelity used in the finite element models of the house structure (e.g., walls, floors, roofs, etc.) may have an impact on these exposure estimates. This presentation documents a recent study in which the fidelity of the structural finite element models was varied. Model fidelity was either an orthotropic panel approximation, in which the stiffening effects of studs were smeared over the entire panel, or a model that explicitly modeled the sheathing and studs. For sonic boom noise, it was found that the orthotropic panel approximation performs well for partitions that have through-the-thickness geometric symmetry, for example walls with two sheathing surfaces. However, the orthotropic approximation does not perform as well for panels with only one sheathing surface, which is typical of floors, ceilings, and roofs.

4:00

2pNSb8. Sonic boom weather analysis of the F-18 low boom dive maneuver. Juliet Page (Environ. Measurement and Modeling, Volpe National Transportation Systems Ctr., 55 Broadway, Cambridge, MA 02142, juliet.page@dot.gov)

In support of community low boom test planning, a sonic boom analysis of ten years of weather data was conducted at multiple coastal regions for an F-18 conducting the NASA low boom dive maneuver. The low boom dive maneuver involves an inverted dive where the aircraft accelerates supersonically and then pulls out above 30,000 Ft. During the dive maneuver, the sonic booms arrive on both egg and crescent shaped isopemps. Due to the supersonic flight conditions and the propagation paths the boom from the earlier parts of the trajectory arrive before the later part of the flight path. The influence of the local meteorological conditions on this maneuver has a striking effect on the sonic boom footprints, including the shape and location of the focal zone and the extent of the low-amplitude sonic boom carpet region. The paper will describe the PCBoom sonic boom propagation results and interpretive techniques for assessing potential coastal sites for conducting dose-response testing using the F-18 dive maneuver.

4:20

2pNSb9. A study of reflected sonic booms using airborne measurements. Samuel R. Kantor and Larry J. Cliatt (AeroDynam. and Propulsion, NASA Armstrong Flight Res. Ctr., 4800 Lily Ave., Edwards, CA 93523, samuel.r.kantor@nasa.gov)

In support of ongoing efforts to bring commercial supersonic flight to the public, the Sonic Booms in Atmospheric Turbulence (SonicBAT) flight test was conducted at NASA Armstrong Flight Research Center. During this test, airborne sonic boom measurements were made using an instrumented TG-14 motor glider, called the Airborne Acoustic Measurement Platform (AAMP). During the flight program, the AAMP was consistently able to measure the sonic boom wave that was reflected off of the ground, in addition to the incident wave, resulting in the creation of a completely unique data set of airborne sonic boom reflection measurements. This paper focuses on using this unique data set to investigate the ability of sonic boom modelling software to calculate sonic boom reflections. Because the algorithms used to model sonic boom reflections are also used to model the secondary carpet and over the top booms, the use of actual flight data is vital to improving the understanding of the effects of sonic booms outside of the primary carpet. Understanding these effects becomes especially important as the return of commercial supersonic approaches, as well as ensuring the accuracy of mission planning for future experiments.

4:40

2pNSb10. The minimum number of ground measurements required for narrow sonic boom metric confidence intervals. William Doebler and Victor Sparrow (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, wfd5057@psu.edu)

In subsonic civilian flight, the FAA's permissible noise standard sets a limit on average aircraft loudness and requires that the loudness is known to within an adequately narrow 90% confidence interval. The FAA and international partners are developing a certification standard for the enroute regime of overland civil supersonic flight. In support of developing this standard, it may be useful to identify the number, location, and frequency of ground measurements to ensure that metrics' 90% confidence intervals for a supersonic flyover are acceptably narrow. Using NASA's Superboom Caustic Analysis and Measurement Program (SCAMP) database where an F-18 jet flew above a linear 3048 m long 81-microphone array, confidence intervals of array-averaged metric values were calculated for six steady speed, level altitude flights. Microphones were selectively removed from the metric averaging process using various techniques to identify the effect of microphone number on confidence interval size. Preliminary results indicate ten measurements yield sufficiently narrow confidence intervals compared to a large number of measurements. [Work supported by the FAA. The opinions, findings, conclusions, and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

5:00

2pNSb11. Commercial space operations noise and sonic boom issues. Natalia Sizov (Office of Environment and Energy, Federal Aviation Administration, 800 Independence Ave., SW, Washington, DC 20591, natalia.sizov@faa.gov)

Commercial space transportation is a rapidly developing industry worldwide. The expansion of the space transportation infrastructure creates additional challenges to the National Airspace System. New launch facilities have and are being developed some of which co-locate with commercial airports. Rocket launch community noise impact requires adequate assessment and mitigation. New space vehicles encompass a wide range of design, aircraft geometry, and flight parameters. A database of operational launch profiles for rockets does not exist at this moment. The acoustical characteristic of these new vehicles may also differ from that of existing rockets or conventional aircraft. Spacecraft ascent sonic boom signatures have higher overpressure, longer duration and a bow shock. There are no standard methodologies for the environmental assessment of launch vehicles and sites. Developing and validating these models is an emerging field. In addition, the metrics used for community noise assessment for commercial aircraft are being used for commercial space operations and may not be appropriate for such use. The existing gaps in rocket launch community noise assessment, technical and regulatory requirements, and current steps the FAA is undertaking to establish an environmental regulatory framework for the commercial space operations will be discussed.

2p MON. PM

2pPA

Physical Acoustics: Infrasound II

Roger M. Waxler, Cochair

NCPA, University of Mississippi, 1 Coliseum Dr., University, MS 38677

Pieter Smets, Cochair

*R&D Department of Seismology and Acoustics, KNMI, PO Box 201, De Bilt 3730 AE, Netherlands**Invited Papers*

1:20

2pPA1. Mapping nonlinear infrasound penetration into a shadow zone: Results from rocket motor blasts at the Utah Test and Training Range. Catherine de Groot-Hedlin (Scripps Inst. of Oceanogr., Univ. of California at San Diego, 9500 Gilman Dr., La Jolla, CA 92037-0225, chedlin@ucsd.edu)

Each summer, large-scale detonations are carried out at the Utah Test and Training Range (UTTR) west of Salt Lake City. In 2016, acoustic sensors were placed at ranges up to 90 km east of the UTTR test site. The frequencies of the recorded signals indicate that they are infrasonic waves. The travel times imply that they are direct arrivals although the sensors lie within a shadow zone. Finally, the signal amplitudes indicate that acoustic propagation is nonlinear near the source. Consequently, numerical simulations that account for diffraction, and the effects of strong shocks are required to accurately map infrasound propagation into this region. A finite difference time-domain infrasound propagation method is applied to these signals. The algorithm relies on the assumption that the environmental model is azimuthally symmetric about the source location, allowing for efficient numerical computation of acoustic propagation from a spherical source. For each detonation, numerical computations are performed along a series of azimuths from the shot position, using accurate weather and topography along each path. The results show that infrasound penetration into the shadow zone is accurately predicted. The synthesized over-pressures are positive correlated with observed pressure amplitudes, to an accuracy of 8 dB.

1:40

2pPA2. A numerical study of infrasound scattering from atmospheric inhomogeneities based on the 3-D unsteady compressible Navier-Stokes equations. Roberto Sabatini (Ctr. for Acoust., Ecole Centrale de Lyon, 36 Ave. Guy de Collongue, Ecully cedex 69134, France, roberto.sabatini@doctorant.ec-lyon.fr), Olivier Marsden (ECMWF, Reading, United Kingdom), Christophe Bailly (Ctr. for Acoust., Ecole Centrale de Lyon, Ecully, France), and Olaf Gainville (CEA/DAM/DIF, Arpajon, France)

A direct numerical simulation of the compressible unsteady Navier-Stokes equations is performed to investigate the 3-D nonlinear acoustic field generated by a high-amplitude infrasonic source placed at ground level in a realistic atmosphere. High-order finite differences and a Runge-Kutta time integration scheme originally developed for aeroacoustic applications are employed. The atmosphere is parametrized as a stationary and vertically stratified medium, constructed by specifying a speed of sound and a mean wind profiles which mimic the main trends observed during the Misty-Picture experiment. In the present talk, after a general description of the acoustic field observed up to 140 km altitude and 450 km range, the scattering from stratospheric inhomogeneities is investigated. The spectrum of the scattered wave recorded at ground level is more particularly discussed and its dependence on the spectral properties of the inhomogeneities is highlighted. A fast method for computing the scattered field, based on a wavelet representation of the temperature and wind fluctuations, is finally presented.

2:00

2pPA3. Local infrasound propagation in three dimensions simulated with *in situ* atmospheric measurements. Keehoon Kim, Arthur Rodgers (Geophysical Monitoring Program, Lawrence Livermore National Lab., 7000 East Ave., L-103, Livermore, CA 94550, kim84@llnl.gov), and Douglas Seastrand (Remote Sensing Lab., National Security Technologies, Las Vegas, NV)

Local infrasound propagation is influenced by atmospheric conditions. The vertical gradients of local ambient temperatures and winds can alter the effective sound speed profiles in the atmosphere and dramatically change the focusing and defocusing behaviors of acoustic waves at local distances. Accurate prediction of local infrasound amplitude is critical to estimating explosion energies of natural and/or man-made explosions, and physics-based numerical simulation that accounts for three-dimensional propagation effects should be required for that purpose. The accuracy of a numerical modeling is, however, often compromised by the uncertainty of atmospheric parameters that are used for the modeling. In this study, we investigate the impacts of local atmospheric conditions on infrasound propagation using the data from chemical explosion experiments. *In situ* atmospheric conditions during the experiments are measured by a combination of (1) local radiosonde soundings, (2) Atmospheric Sounding Spectrometer for Infrared Spectral Technology (ASSIST), (3) surface weather stations, and (4) a wind LIDAR profiler, which can complement atmospheric profiles for numerical simulations and capture local atmospheric variability. We simulate three-dimensional local infrasound propagation using a finite-difference method with the local atmospheric measurements, and the accuracy of the numerical simulations are evaluated by the comparison with the field observations.

2:20

2pPA4. Estimating the effects of an ellipsoidal earth and topography on infrasonic propagation. Philip Blom (Los Alamos National Lab., Los Alamos National Lab., PO Box 1663, Los Alamos, NM 87545, pblom@lanl.gov)

Infrasonic signals often propagate significant horizontal distances so that predictions obtained using a flat ground approximation can introduce inaccuracies. Simulations of propagation in an atmospheric layer around a spherical globe have shown non-negligible deviations from flat ground predictions for both arrival locations and propagation times. Simulation predictions for flat-ground and spherical earth models will be discussed and the additional challenges of implementing a non-spherical globe model and the inclusion of topography will be discussed using the approximation of geometric acoustics. A non-spherical globe model, such as the WGS84 ellipsoid, is found to produce range dependence in the propagation medium, even in the case that a stratified local atmosphere is assumed. Further, although scattering and diffraction effects are not included in the geometric limit, variations in the ground surface level can be included in ray path computation to more accurately model propagation of infrasound. Propagation effects will be detailed in the case of a tropospheric waveguides for which interaction with the ground surface is significant as well as the case that source and receiver locations have differences in elevation.

2:40

2pPA5. Local-distance acoustic propagation from explosions. Stephen Arrowsmith, Nathan Downey, Leiph Preston, and Daniel C. Bowman (Sandia National Labs., PO Box 5800, Albuquerque, NM 87185-0404, sjarrow@sandia.gov)

We study the effect of acoustic propagation from explosions on full waveforms using both empirical and numerical approaches. Empirically, we explore the effects of meteorology, terrain, etc., on explosion signatures by exploiting a rich dataset of explosion measurements in different regions to relate specific path effects to second-order effects in the waveforms. Numerically, we explore the effects using different full wave codes to understand observations from a unique experiment with both ground and air waveform and 3D wind field measurements. We discuss implications for explosion yield estimation for surface explosions and for underground events.

Contributed Papers

3:00

2pPA6. The study of sudden stratospheric warmings using infrasound. Pieter Smets, Jelle Assink, and Láslo Evers (R&D Dept. of Seismology and Acoust., KNMI, PO Box 201, De Bilt 3730 AE, Netherlands, smets@knmi.nl)

Infrasound has a long history in monitoring SSWs. Several pioneering studies have focused on the various effects of a major warming on the propagation of infrasound, described throughout this chapter. A clear transition can be denoted from observing anomalous signatures towards the use of these signals to study anomalies in upper atmospheric specifications. First studies describe the various infrasonic signatures of a major warming. In general, the significant change in observed infrasound characteristics correspond to summer-like conditions in midwinter. More subtle changes are denoted during a minor warming, recognizable by the presence of a bidirectional stratospheric duct. A combined analysis of all signal characteristic unravels the general stratospheric structure throughout the life cycle of the warming. From then on, infrasound observations are used to evaluate the state of the atmosphere as represented by various NWP models. A new methodology, comparing regional volcano infrasound with simulations using various forecast steps, indicates interesting variations in stratospheric skill.

3:20–3:40 Break

3:40

2pPA7. NCPAprop—A software package for infrasound propagation modeling. Roger M. Waxler (National Ctr. for Physical Acoust., Univ. of MS, University, MS), Jelle D. Assink (The Royal Netherlands Meteorological Inst., De Bilt, Netherlands), Claus Hetzer (National Ctr. for Physical Acoust., Univ. of MS, University, MS), and Doru Velea (Leidos, 14668 Lee Rd., Chantilly, VA 20151, doru.velea@leidos.com)

Developed by the infrasound group at the National Center of Physical Acoustics, University of Mississippi, and a few collaborators, NCPAprop is an open source software package aiming at providing a comprehensive set of tested and validated numerical models for simulating the long range propagation of infrasonic signals through the earth's atmosphere. The algorithms implemented in NCPAprop are designed for frequencies large enough that

the effects of buoyancy can be neglected and small enough that propagation to ranges of hundreds to thousands of kilometers is possible without significant signal attenuation. Nominally, NCPAprop can, without modification, be used to efficiently model narrowband propagation from 0.1 to 10 Hz and broadband propagation from 0.05 Hz to 2 or 3 Hz. NCPAprop provides both geometrical acoustic and full wave models which will be presented. The geometrical acoustics part consists of 2-d and 3-d ray tracing programs as well as a non-linear ray theory model. The full wave models consist of a suite of normal mode models of increasing complexity and a Parabolic Equation (PE) model.

4:00

2pPA8. Acoustic/Infrasonic analysis and modeling of thunder from a long-term recording in Southern France. Arthur Lacroix (Institut Jean Le Rond d'Alembert (UMR 7190), Univ. Pierre and Marie Curie - Paris 6, Université Pierre et Marie Curie, 4 Pl. Jussieu, Paris 75005, France, arthur.lacroix@upmc.fr), Thomas Farges (CEA, DAM, DIF, Arpajon, France), Régis Marchiano, and François Coulouvrat (Institut Jean Le Rond d'Alembert (UMR 7190), Univ. Pierre and Marie Curie - Paris 6, Paris, France)

Thunder produces complex signals with a rich infrasonic and audible frequency spectrum. These signals depend both on the source and the propagation to the observer. However, there is no mutual agreement on the link between the observed spectral content and the generation mechanisms. The objectives of this study is to provide additional experimental and theoretical investigations, especially on the return stroke, based on a database of several thousands of acoustic and electromagnetic signals recorded in Southern France during autumn 2012 (HyMeX campaign). It contains a sufficient number of events close to the source (<1 km) to minimize propagation effects and to focus on the source effects. Source localization and lightning acoustical reconstruction indicate that infrasonic and low frequency audible part (1–40 Hz) of the spectrum show no clear differences between the return stroke and the intracloud discharges. These observations are compatible with a source mechanism due to the thermal expansion associated to the sudden heating of the air in the lightning channel. An original model inspired by Few's string pearl theory has been developed. It shows that the tortuous channel geometry explains at least partly the low frequency content of observed thunder spectrum.

2p MON. PM

2pPA9. Infrasound scattering from stochastic gravity wave packets. Christophe MILLET (CEA, DAM, DIF, CEA, DAM, DIF, Arpajon 91297, France, christophe.millet@cea.fr), Bruno RIBSTEIN (LMD, ENS, Cachan, France), and Francois LOTT (CMLA, ENS, Paris, France)

Long-range infrasound propagation problems are characterized by a large number of length scales and a large number of propagating modes. In the atmosphere, these modes are confined within waveguides causing the sound to propagate through multiple paths to the receiver. In most infrasound modeling studies, the small scale fluctuations are represented as a “frozen” gravity wave field that is superimposed on a given average background state, and the normal modes are obtained using a single calculation. Direct observations in the lower stratosphere show, however, that the gravity wave field is very intermittent, and is often dominated by rather well defined large-amplitude wave packets. In the present work, we use a few proper modes to describe both the gravity wave field and the acoustic field. Owing to the disparity of the gravity and acoustic length scales, the acoustic field can be constructed in terms of asymptotic expansions using the method of multiple scales. The amplitude evolution equation involves random terms that can be related to vertically distributed gravity wave sources. To test the validity of the theory, numerical results are compared with recorded signals. It is shown that the present stochastic theory offers significant improvements over current semi-empirical approaches.

4:40

2pPA10. Spectral broadening of infrasound tones in mountain wave fields. Florentin DAMIENS (LMD, ENS, Paris, France), Christophe MILLET (CEA, DAM, DIF, CEA, DAM, DIF, Arpajon 91297, France, christophe.millet@cea.fr), and Francois LOTT (LMD, ENS, Paris, France)

Linear theory of acoustic propagation is used to analyze how infrasounds trapped within the lower tropospheric waveguide propagate across mountain waves. The atmospheric disturbances produced by the mountains are predicted by a semi-theoretical mountain gravity wave model. For the infrasounds, we solve the wave equation under the effective sound speed approximation both using a spectral collocation method and a WKB approach. It is shown that in realistic configurations, the mountain waves can deeply perturb the low level waveguide, which leads to significant acoustic dispersion. To interpret these results, we follow each acoustic mode separately and show which mode is impacted and how. We show that during statically stable situations, roughly representative of winter or night situations, the mountain waves induce a strong Fohn effect downstream which shrinks significantly the waveguide. This yield a new form of infrasound absorption, a form that can largely outweigh the direct effect of the mask the mountain induce on the low-level waveguide. On the opposite, when the low level flow is less statically stable (summer or day situations), the mountain wave dynamics does not produce dramatic responses downstream, it can even favor the passage of infrasound waves, somehow mitigating the direct effect of the obstacle.

2pPA11. Simulating global atmospheric microbaroms from 2010 onward. Pieter Smets, Jelle Assink, and Láslo Evers (R&D Dept. of Seismology and Acoust., KNMI, PO Box 201, De Bilt 3730 AE, Netherlands, smets@knmi.nl)

Microbaroms are atmospheric pressure oscillations radiated from non-linear ocean surface wave interactions. Large regions of interacting high-energetic ocean waves, e.g., ocean swell and marine storms, radiate almost continuously acoustic energy. Microbaroms dominate the infrasound ambient noise field, which makes them a preferred source for passive atmospheric probing. Microbarom are simulated using a two-fluid model, representing an atmosphere over a finite-depth ocean and a coupled ocean-wave model providing the sea state. Air-sea coupling is crucial due to the two-way interaction between surface winds and ocean waves. In this study, a detailed overview is given on how global microbarom simulations are obtained, including a sensitivity analysis of the various model input data and parameterizations. Simulations are validated by infrasound array observations of the International Monitoring Systems (IMS) of the Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO). A brief demonstration is given on the added value of global microbarom simulations for infrasound studies and how to obtain these source simulations.

5:20

2pPA12. Infrasound sensing on Mars: Wind noise predictions for a porous dome geometry. Kevin Pitre and Andi Petculescu (Univ. of Louisiana at Lafayette, University of Louisiana at Lafayette, Lafayette, LA 70503, kmp7935@gmail.com)

Infrasound sensing will play an important role in future Mars exploration. Applications include quantifying bolide impacts, monitoring subsurface activity, storm and dust-devil dynamics, as well as characterizing the planetary boundary layer. As on Earth, infrasonic measurements are likely to be hampered by wind-generated noise at the frequencies of interest. Instead of the rosette-type filter geometry commonly used at Earth monitoring stations, porous hemispherical domes could be more easily deployed in an “inverted-umbrella” configuration, with the sensor at the apex. By adapting previous work (Noble *et al.*, Proc. Meet. Acoust. 21, 045005 (2014)) to the conditions in the Martian surface layer, we predict the infrasonic wind noise at the center of a porous dome placed at the locations of various Mars landers (Viking 1 and 2, Pathfinder, Mars Science Laboratory, and Phoenix). The predictions include the turbulence-turbulence, turbulence-shear, and stagnation-pressure contributions, obtained for different dome porosities and Martian wind speeds. Measurements at Mars’ surface as well as interpolated data from the Mars Climate Database (www-mars.lmd.jussieu.fr), a detailed Mars circulation model, are used as inputs to the model. The work was funded by a grant from the Louisiana Space Consortium.

2pPPa

Psychological and Physiological Acoustics: Models and Reproducible Research II

Alan Kan, Cochair

University of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705

Piotr Majdak, Cochair

*Acoustics Research Institute, Austrian Academy of Sciences, Wohllebengasse 12-14, Wien 1040, Austria**Invited Papers*

1:20

2pPPa1. Development and dissemination of computational models for physiology and psychophysical predictions. Laurel H. Carney (Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu)

This talk will present lessons from our experience in publishing and sharing computational models for physiological and psychophysical responses. Our physiological models have included rather comprehensive and nonlinear phenomenological descriptions of auditory-nerve responses to complex sounds, and simpler linear models for brainstem and midbrain responses. Using ensembles of single-neuron models to estimate population responses enables psychophysical predictions based on different aspects of sub-cortical representations. Examples of psychophysical models that we have pursued include level discrimination and diotic and dichotic masked detection of tones. Some of the challenges inherent in this type of work will be discussed. [Work supported by NIH-R01-001641 & -010813.]

1:40

2pPPa2. From physiology to functional auditory-nerve models: Challenges and approaches. Sarah Verhulst (Ghent Univ., Technologiepark 15, Zwijnaarde 9052, Belgium, s.verhulst@ugent.be)

A variety of auditory-nerve models, as well as a vast amount of animal single-unit and population response data that can be used to set the parameters of such models, exists. However, it is hard to evaluate different model implementations from published data to decide whether the specific implementation is appropriate for your envisioned application. In this presentation, I will give an experience-based overview on the challenges faced with when evaluating the model parameters in auditory-nerve models. The adopted approach uses the available computer code of the different models under test, and compares their responses to the same input. This method is very efficient in testing the influence of changing one specific part of the model while leaving the rest unchanged, and can ultimately yield improved functional models of the auditory periphery.

2:00

2pPPa3. Simulation model for interaural time difference discrimination for tones. William M. Hartmann (Phys. and Astronomy, Michigan State Univ., Physics-Astronomy, 567 Wilson Rd., East Lansing, MI 48824, hartmann@pa.msu.edu) and Andrew Brughera (Hearing Res. Ctr., Boston Univ., Boston, MA)

Difference limens for the interaural time difference (ITD) can be measured with a 2-interval, 2-alternative forced-choice staircase using the ITD as the staircase variable. Experimental results can be predicted by a computational model that simulates the experiment protocol in every important detail, as applied to human listeners, but the computer can tolerate hundreds of times more runs. At the core of the simulation is a decision process based on an opponency model for the two medial superior olives (MSO). MSO firing rates as functions of ITD are initially determined by a stochastically driven Hodgkin-Huxley cell model and represented in the simulation by a four-parameter fitted function. A corresponding noise function is estimated from multiple runs of the cell model. Left-right symmetry in both the model and the experiment protocol simplifies the calculations. Simulations have practical value in relating staircase thresholds to the underlying parameterized firing rate functions for given staircase variables, especially the initial ITD and the step size. Understanding this relationship is critical for the design and evaluation of experiments at low tone frequencies where thresholds grow to span a wide range. [Work supported by the AFOSR and ARCLP.]

2:20

2pPPa4. Reproducing response characteristics of electrically-stimulated auditory nerve fibers with a phenomenological model. Marko Takanen and Bernhard U. Seeber (Audio Information Processing, Tech. Univ. of Munich, Arcisstrasse 21, Munich 80333, Germany, marko.takanen@tum.de)

Electrical stimulation of the auditory nerve fibers (ANFs) by a cochlear implant (CI) restores hearing for profoundly deaf people. Modern CIs use sequences of amplitude-modulated charge-balanced pulses to encode the spectro-temporal information of the sound reaching the ears of the listener. In such a pulsatile stimulation, several temporal phenomena related to inter-pulse interactions affect the

responsiveness of the ANF during the course of the stimulation. Specifically, refractoriness, facilitation, accommodation, and spike-rate adaptation affect whether a given pulse evokes an action potential or not, and these phenomena continue to provide challenges for computational models. Here, we present a model that builds on recent the biphasic leaky integrate-and-fire model by Horne *et al.* (Front. Comput. Neurosci. 2016), which we have extended to include elements that simulate refractoriness and facilitation/accommodation by affecting the threshold value of the model momentarily after supra- and subthreshold stimulation, respectively. We show that the revised model can reproduce neurophysiological data from single neuron recordings considering the aforementioned phenomena. By accurate modeling of temporal aspects of inter-pulse interactions, the model is shown to account also for effects of pulse rate on the synchrony between the pulsatile input and the spike-train output. [Work supported by BMBF 01 GQ 1004B.]

2:40

2pPPa5. Modeling sound externalization based on listener-specific spectral cues. Robert Baumgartner (Acoust. Res. Inst., Austrian Acad. of Sci., Wohllebengasse 12-14, Vienna 1040, Austria, robert.baumgartner@oeaw.ac.at), Piotr Majdak (Acoust. Res. Inst., Austrian Acad. of Sci., Wien, Austria), H. Steven Colburn, and Barbara Shinn-Cunningham (Dept. of Biomedical Eng., Boston Univ., Boston, MA)

Sound sources in natural environments are usually perceived as externalized auditory objects located outside the head. In contrast, when listening via headphones or hearing-assistive devices, sounds are often heard inside the head, presumably because they are filtered in a way inconsistent with normal experience. Previous results suggest that high-frequency spectral cues arising from the listener-specific filtering by the pinnae are particularly important for sound externalization, but this has not been confirmed in a quantitative perceptual model yet. Here, we present a model designed to predict sound externalization related to the spectral-cue salience in free field. The modeling results are compared to results from various behavioral studies testing the effect of low-pass filtering, non-individualized head-related transfer functions, and behind-the-ear microphone casing in hearing-assistive devices. We will discuss the limitations of previous experimental designs and existing modeling approaches, including fundamental issues of model fitting.

3:00

2pPPa6. Predicting binaural lateralization and discrimination using the position-variable model. Richard M. Stern (Elec. and Comput. Eng., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, rms@cs.cmu.edu)

The position-variable model was developed as a means to characterize and predict a variety of binaural lateralization, discrimination, and detection phenomena. The model was motivated by a desire for a more complete understanding of the putative mechanisms by which interaural time and intensity differences were combined, as well as the extent to which results in interaural discrimination and binaural detection experiments are mediated by cues based on subjective lateral position. This paper will describe recent efforts to unify and extend the predictions of the model, as well as to develop a publicly accessible version of the model within the framework for comparing evaluating binaural models described by Dietz *et al.* in this session. Predictions of the model are based on computation of the centroid along the internal-delay axis of the patterns of activity of the display of information proposed earlier by Jeffress and Colburn, derived from the auditory-nerve response to the experimental stimuli. Some of the issues to be discussed include comparisons to other proposed methods of developing lateralization predictions, the impact of internal versus external noise in the model's predictions, and specific issues involved with modifying the model to render it compatible to the common framework for model comparison.

3:20–3:40 Break

3:40

2pPPa7. Reproducible psychoacoustic experiments and computational perception models in a modular software framework. Stephan D. Ewert (Medizinische Physik and Cluster of Excellence Hearing4All, Universität Oldenburg, Carl-von-Ossietzky Str. 9-11, Oldenburg 26129, Germany, Stephan.ewert@uni-oldenburg.de) and Torsten Dau (Hearing Systems Group, Dept. of Elec. Eng., Tech. Univ. of Denmark, Lyngby, Denmark)

Psychoacoustic experiments and auditory models are fundamental elements of hearing research helping to understand human auditory perception. One successful way to apply models has been to use the model as artificial observer, performing exactly the same psychoacoustic experiment as human subjects [e.g., Jepsen *et al.*, J. Acoust. Soc. Am. 124, 422 (2008)]. While the signal processing parts of this and other models are publicly available, reproducible research requires availability of the complete framework including stimulus generation, experimental procedure, and interface to the model. For this, AFC for Matlab/Octave [www.aforcedchoice.com] provides a free and highly flexible framework to design and run psychoacoustic measurements with subjects and computer models. Previous versions of AFC have been used for nearly two decades in several highly ranked psychoacoustic research sites. To foster reproducible research, AFC offers full downward compatibility to the very first version, and the ability to easily overload or add measurement procedures, audio drivers, and models/model interfaces. Here a new version is presented with the above model as a use case. A database of psychoacoustic experiments from numerous publications is established, to provide the stimulus generation of the experiments, methods, and models for exact reproduction of the original work for teaching and research.

4:00

2pPPa8. An initiative for testability and comparability of binaural models. Mathias Dietz (National Ctr. for Audiol., Western Univ., 1201 Western Rd., London, ON N6G 1H1, Canada, mdietz@uwo.ca), Torsten Marquardt (UCL Ear Inst., London, United Kingdom), Piotr Majdak (Oesterreichische Akademie der Wissenschaften, Wien, Austria), Richard M. Stern (Carnegie Mellon Univ., Pittsburgh, PA), William M. Hartmann (Michigan State Univ., East Lansing, MI), Dan F. Goodman (Imperial College, London, United Kingdom), and Stephan D. Ewert (Universitaet Oldenburg, Oldenburg, Germany)

A framework aimed at improving the testability and comparability of binaural models will be presented. The framework consists of two key elements: (1) a repository of testing software that evaluates the models against published data and (2) a model repository. While the framework is also intended for physiological data, the planned initial contribution will be psychoacoustical data together with their

psychoacoustical testing protocols, as well as existing binaural models from available auditory toolboxes. Researchers will be invited to provide their established as well as newly developed models in whatever programming language they prefer, given the models are compatibility with the proposed interface to the testing software. This entails that the models act as artificial observers, testable with exactly the same procedure as the human subjects. A simple communication protocol based on wav and txt-files is proposed because these are supported by every programming environment, and are able connect models and testing software of any programming language. Examples will illustrate the principle of testing models with unaltered signal processing stages on various seminal data sets such as tone detection in so-called double-delayed masking noise, or lateralization of $\frac{3}{4}$ -period delayed noise and sounds with temporally asymmetric envelopes.

4:20

2pPPa9. Resource sharing in a collaborative study on cochlear synaptopathy and suprathreshold-hearing deficits. Hari M. Bharadwaj, Jennifer M. Simpson, and Michael G. Heinz (Speech, Lang., and Hearing Sci. & Biomedical Eng., Purdue Univ., 715 Clinic Dr., Lyles-Porter Hall, West Lafayette, IN 47907, hbharadwaj@purdue.edu)

Evidence from animal models of substantial noise-induced cochlear synaptopathy even in the absence of measurable audiometric changes has led to an active debate over whether such damage occurs in humans, and whether it contributes to suprathreshold-hearing deficits. Addressing these fundamental and translational questions requires multi-disciplinary approaches that integrate widely ranging forms of data and analyses, e.g., animal/human/model, evoked/single-unit/behavioral, and lab/clinical. Furthermore, connecting results across research groups around the world working on various species requires a systematic approach to resource sharing that will promote rigor and reproducibility. Here, we describe our efforts and plans to share resources from a large-scale collaborative project on noise-induced synaptopathy that links single-unit, evoked, and behavioral data from chinchillas with evoked, behavioral, and imaging data from humans studied in the laboratory and in the clinic. In addition to using modular implementations of stimulation paradigms, computational models, and analysis tools in high-level languages, we adopt open-access resource repositories and integrated platform-independent tools for version control, distributed development, documentation, and testing. Such resource sharing will help expedite answering the question of whether the anatomical/physiological effects seen in smaller animal models are present and perceptually significant in humans.

4:40

2pPPa10. Open community platform for hearing aid algorithm research. Hendrik Kayser (Medizinische Physik and Cluster of Excellence H4a, Carl von Ossietzky Universität Oldenburg, Ammerlaender Heerstrasse 114-118, Oldenburg D-26111, Germany, hendrik.kayser@uol.de), Tobias Herzke, Frasher Loshaj (HörTech gGmbH, Oldenburg, Germany), Giso Grimm, and Volker Hohmann (Medizinische Physik and Cluster of Excellence H4a, Carl von Ossietzky Universität Oldenburg, Oldenburg, Germany)

The project “Open community platform for hearing aid algorithm research” funded by the National Institutes of Health (NIH Grant 1R01DC015429-01) aims at sustainable, focused research toward improvement and new types of assistive hearing systems. To this end, an open-source software platform for real-time audio signal processing will be developed and made available to the research community including a standard set of reference algorithms. Furthermore, novel algorithms for dynamic and frequency compression, auditory-scene-analysis based noise suppression and speech enhancement, and feedback management will be investigated. For a realistic assessment of the benefits of hearing aid algorithms and combinations thereof, instrumental measures of performance in virtual acoustic environments of varying complexity will be included in the algorithm design and optimization. With such a quasi-standard set of benchmarks and the means to develop and integrate own signal-processing methods and measures in the same framework, the platform enables reproducible, comparative studies and collaborative research efforts. Beyond an implementation for PC hardware the system will also be made usable for ARM-processor based hardware to allow pre-development of wearable audio devices—so-called “hearables.” This contribution will present underlying previous work and the goals and plans of the project that has started midyear 2016. www.openMHA.org.

5:00

2pPPa11. Open science in the Two!Ears project—Experiences and best practices. Hagen Wierstorf (Audiovisual Technol. Group, Technische Universität Ilmenau, Ehrenbergstraße 29, Ilmenau 98693, Germany, hagen.wierstorf@posteo.de), Fieta Winter, and Sascha Spors (Inst. of Communications Eng., Univ. Rostock, Rostock, Germany)

Two!Ears was an EU funded project for binaural auditory modeling with ten international partners involved. One of the project goals was to follow an Open Science approach in all stages. This turned out to be a challenging task as the project involved huge amounts of software, acoustical measurements, and data from listening tests. On the other hand, it was obvious from the positive experience with the Auditory Modelling Toolbox that an Open Science approach would have a positive impact and foster progression afterwards. As there existed no ready solution to achieve this goal at the beginning of the project, different paths for data management were tested. It was especially challenging to provide a solution for data storage. Here, the goal was not only the long term accessibility of the data, but also the revision control of public and private data for the development inside the project. In the end, the project was able to make most of its software and data publicly available, but struggled to apply the reproducible research principle to most of its papers. This contribution will discuss best practices to actively support reproducible research in large-scale projects in the acoustic community, points out problems and solutions.

5:20

2pPPa12. A library of real-world reverberation and a toolbox for its analysis and measurement. James Traer and Josh McDermott (Brain and Cognit. Sci., MIT, 77 Massachusetts Ave., Cambridge, MA 02139, jtraer@mit.edu)

Reverberation distorts the sounds produced in the world, but in doing so provides information about the environment. This distortion is characterized by the Impulse Response (IR) and depends upon the material and geometry composing the environment. For some tasks (voice recognition, source localization, acoustic tomography, etc.), reverberation must be discounted, but for others (room identification,

distance estimation, etc.), it must be analyzed. Our recent work on the perception of reverberation has leveraged measurements of real-world IRs, which exhibit a number of regularities that the brain appears to have internalized for auditory scene analysis. Here, we present a library of these measurements and a toolbox for measuring and analyzing additional IRs. The library contains 271 IRs from spaces encountered by 7 volunteers over 2 weeks of daily life, and thus reflects the distribution of typical reverberation experienced by humans. The toolbox includes procedures for measuring IRs with low-cost, portable equipment, with a low-volume broadcast. This allows measurement in both public and outdoor spaces. Both the library and toolbox are publicly available.

Contributed Paper

5:40

2pPPa13. The open virtual auditory-localization environment: Towards a common methodology for objectively evaluating head-related transfer function personalization methods. Griffin D. Romigh (Air Force Res. Labs, 2610 Seventh St., Area B, Bldg. 441, Wright Patterson AFB, OH 45433, griffin.romigh@us.af.mil) and Jason Ayers (Ball Aerosp., Dayton, OH)

Despite the fact that individualized head-related transfer functions (HRTFs) are critical for achieving high-fidelity virtual audio representation, the techniques for measuring them, which have been around for decades, are too costly for most potential users. As such, many strategies have been proposed that aim to improve virtual audio fidelity by personalizing a non-individualized HRTF based on input (or anthropometric information) from

the user. Unfortunately, evaluations of these methodologies have varied widely from purely computational to purely subjective, making comparisons across studies or to objective behavioral performance metrics difficult. The current work presents the Open Virtual Auditory-Localization Environment (OpenVALE), a software toolkit for providing a common, objective, HRTF-based auditory localization task via new, relatively low-cost, commercial VR headsets. The heart of OpenVALE is a server application that allows researchers to dynamically load custom HRTFs, present spatialized auditory stimuli, collect hand- or head-slaved, cursor-based localization responses, and provide visual feedback, all through simple string-based IP socket messages from any compatible client application (e.g., Matlab, Java, Python, etc.). An initial validation of the task environment, based on individualized HRTF measurements, will be described along with a discussion of the remaining challenges for creating an accepted standard methodology.

MONDAY AFTERNOON, 26 JUNE 2017

BALLROOM B, 1:20 P.M. TO 4:20 P.M.

2pPPb

Psychological and Physiological Acoustics: Hearing Aiding, Protection, and Speech Perception

Valeriy Shafiro, Chair

Communication Disorders & Sciences, Rush University Medical Center, 600 S. Paulina Str., AAC 1012, Chicago, IL 60612

Contributed Papers

1:20

2pPPb1. An investigation of passive type hearing protection in human and animals by their auricles by diverting natural drainage of rain water along facial features into the ear canal. Amitava Biswas (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Many textbooks in human auditory system begins with the external ear and basically describe the auricle as a collector of sound. In this study, we have explored the utility of the auricle to protect the external ear canal from environmental factors such as rain shower or sand storm. A model of human head of typical dimensions was held in upright position inside a bath tub. A shower head was positioned directly above the model head. One ear of the model was sliced off. Each ear canal was provided internally a tube of about 8 mm diameter connected to a collecting bottle of about 500 cc capacity. The shower head output was about eight liters of water per minute. The results suggest that the impaired ear is more vulnerable to water entry compared to the unimpaired ear of the model. Therefore, another vital necessity of evolution of the external ear may have been environmental protection, in addition to collection of sound energy. Generalization of the data for other common animals will be discussed.

1:40

2pPPb2. Auditory protection, observation vs standards. Gerald Fleischer (Justus-Liebig-Univ., Hoehenstr. 18, Giessen 35466, Germany, gerald.fleischer@gmx.net)

A summary of more than two decades of research, related to the relationship between the acoustic environment and the auditory threshold. Special groups have been examined: professional musicians, dentists, fans and avoiders of discotheques, office personnel, nomadic people, etc. Persons who suffered from noise-induced damage have been examined, and the acoustic conditions, mostly impulses, reenacted and analyzed. Depending on the pressure-time-history, impulses show several types of characteristic damage (footprints). These types of damage can be determined automatically, using pattern recognition, if more frequencies are used for audiometry. Evidence for training of the auditory system is widespread. A cochlear mechanism for reducing the sensitivity very rapidly appears likely and is presented and discussed. Middle-ear muscles are responsible for auditory accommodation—listening to events nearby while suppressing noise from a distance. They are not helpful for avoiding typical damages caused by noise. Using these parameters gives a much better understanding what is harmful for hearing. This is helpful for avoiding such conditions. Comparing the auditory threshold between various groups, independent of age, reveals what is good for hearing.

2:00

2pPPb3. Attenuation of dual hearing protection: Measurements and finite-element modeling. Hugues Nélisse, Franck C. Sgard (IRSST, 505 Blvd. De Maisonneuve Ouest, Montreal, QC H3A 3C2, Canada, hugues.nelisse@irsst.qc.ca), Marc-André Gaudreau (Cégep de Drummondville, Montréal, QC, Canada), and Thomas Padois (Mech. Eng., École de Technologie Supérieure (ÉTS), Montréal, QC, Canada)

In extremely noisy environments, it is normally recommended to use a combination of earplug and earmuff, denoted here as a dual protection, to protect workers from the excessive noise. Unfortunately, it has been shown repeatedly that the attenuation values obtained with the dual protection are generally less than the sum of the individual earplug and earmuff attenuation values. In the literature, this is generally explained by the bone conduction path and the coupling between the earplug and the earmuff. However, there is much less work devoted to examining in detail the coupling between the earplug and the earmuff. In this work, experimental results on human subjects and on an artificial text fixture are collected using REAT and MIRE procedures for different combinations of earmuffs and earplugs. Additionally, a finite-element model is used to investigate the physics of the problem and to better understand the nature of the coupling between the earplug and the earmuff when used in a dual configuration. Results from the experimental procedures as well as from the FE model are presented and discussed. These results clearly illustrate the importance of the coupling between the earplug and the earmuff when used in combination.

2:20

2pPPb4. Correlation of flow field and acoustic output from hearing aids under influence of wind. Florian Zenger, Linda Gerstner (Inst. of Process Machinery and Systems Eng., Friedrich-Alexander Univ. Erlangen-Nürnberg, Cauerstr. 4, Erlangen 91058, Germany, ze@ipat.uni-erlangen.de), Alexander Lodermeier (Inst. of Process Machinery and Systems Eng., Friedrich-Alexander Univ. Erlangen-Nürnberg, Erlangen, Bayern, Germany), and Stefan Becker (Inst. of Process Machinery and Systems Eng., Friedrich-Alexander Univ. Erlangen-Nürnberg, Erlangen, Bavaria, Germany)

Wind noise in hearing aids occurs even at low wind speeds and is a confounding factor for hearing aid wearer, hence leading to a reduction of speech intelligibility. In this submission, a study on the correlation of the flow field around a hearing aid to its acoustic output is made. The BTE (behind the ear) hearing aid is mounted on an artificial head with three different ear geometries. The flow field is captured using a two component PIV (particle image velocimetry) system. For exposing critical flow phenomena, a POD (proper orthogonal decomposition) of the PIV measurement data is made. The hearing aid output is measured with a microphone inside the artificial head. On the one hand, wind noise in hearing aids is generated by the fluctuating velocity field of the boundary layer on the hearing aid. On the other hand, based on the PIV data and the POD results, flow patterns around the artificial head and the hearing aid are detected, which cause further noise, that is captured by the hearing aid microphones. With these findings, modifications on the hearing aid geometry are deduced, that lead to a decrease in wind noise and hence to a better speech intelligibility.

2:40

2pPPb5. Measurements of the acoustic feedback path of hearing aids on human subjects. Tobias Sankowsky-Rothe and Matthias Blau (Inst. of Hearing Technol. and Audiol., Jade Hochschule WOE, Ofener Straße 16-19, Oldenburg, Niedersachsen 26121, Germany, Tobias.Sankowsky@jade-hs.de)

Feedback is a problem in hearing aids which will cause signal degradation and reduce the maximum applicable gain. More specifically, the advantages of open fittings (e.g., minimizing the occlusion effect) are limited by acoustic feedback. Feedback cancellation algorithms are used to overcome these limitations. For the development of such algorithms, the acoustic feedback path of the hearing aid must be known. The acoustic feedback path is not only affected by the outer sound field but by the individual anatomy and physiology as well. In order to quantify these different influences, feedback path measurements were performed on 20 human subjects. The measurements included different static conditions as well as dynamic ones (i.e.,

repetitive movements were performed during the measurement). Since the sound pressure level must be limited in measurements on human subjects, a valid identification of the feedback path is difficult in many cases, due to a low signal to noise ratio. Therefore, all measurements were performed reciprocally in addition to the direct measurements. Results show that yaw movements only have a small influence on the acoustic feedback path, whereas changes of the outer sound field can have a substantial impact on the feedback path.

3:00

2pPPb6. How to improve a hearing aid fitting based on idiosyncratic consonant errors. Ali Abavisani and Jont b. Allen (ECE, Univ. of Illinois at Urbana-Champaign, 405 N Mathews Ave., Rm. 2137, Urbana, IL 61801, aliabavi@illinois.edu)

The goal of this study is to quantify a given hearing aid insertion gain using a consonant recognition based measure, for ears having sensorineural hearing loss. The basic question addressed is how a treatment impacts phone recognition, relative to a normal-hearing insertion gain. These tests are directed at (1) fine-tuning a treatment, with the ultimate goal of improving speech perception, and (2) to identify when a hearing level based treatment degrades speech recognition. Eight subjects with hearing loss were tested under two conditions: Flat-gain and a Treatment insertion gain based on subject's hearing level. The speech corpus consisted of consonant-vowel tokens at different signal to speech-weighted noise (SNR) conditions, presented at subjects' most comfortable level. The tokens used in this study were selected from those having less than 3% error at -2 [dB] SNR, from 30 Normal Hearing subjects. The Treatment caused token score to improve for 31% of the trials and decrease for 12%. An analysis method was devised to identify degraded tokens for individual hearing impaired ears, based on sorting the tokens according to their error. By comparing sorted errors across experiments, the effect of the treatment could be accurately evaluated, providing precise characterization of idiosyncratic phone recognition.

3:20

2pPPb7. Consistency and variation in recognition of text and speech interrupted at variable rates. Valeriy Shafiro (Commun. Disord. & Sci., Rush Univ. Medical Ctr., 600 S. Paulina Str., AAC 1012, Chicago, IL 60612, valeriy_shafiro@rush.edu), Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, Columbia, SC), and Kimberly Smith (Speech Pathol. and Audiol., Univ. of South Alabama, Mobile, AL)

Recent research indicates that recognition of interrupted text can predict speech intelligibility under adverse listening conditions. However, factors underlying the relationship between perceptual processing of speech and text are not fully understood. We examined contributions of underlying linguistic and perceptual structure by comparing recognition of printed and spoken sentences interrupted at different rates (0.5–64 Hz) in 14 normal-hearing adults. The interruption method approximated deletion and retention of rate-specific linguistic information across the two modalities by substituting white space for silent intervals. Results indicate a remarkably similar U-shaped pattern of cross-rate variation for both modalities, with minima at 2 Hz. Nevertheless, at high and low interruption rates text recognition exceeded speech recognition, while the reverse trend was observed at middle rates. Surprisingly, no significant correlations were obtained in recognition accuracy between text and speech conditions. These findings indicate a high degree of perceptual constancy in recognition of interrupted text and speech, which may rely on retention of rate-specific linguistic and perceptual information retained after the interruptions. On the other hand, results also indicate rate-specific variation in perceptual processing of text and speech, which may potentially affect the degree to which recognition accuracy in one modality is predictive of the other.

3:40

2pPPb8. Predicting consonant recognition and confusions using a microscopic speech perception model. Johannes Zaar and Torsten Dau (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørstedes Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, jzaar@elektro.dtu.dk)

The perception of consonants has been investigated in various studies and shown to critically depend on fine details in the stimuli. The present

study proposes a microscopic speech perception model that combines an auditory processing front end with a correlation-based template-matching back end to predict consonant recognition and confusions. The model represents an extension of the auditory signal processing model by Dau *et al.* [(1997), *J. Acoust. Soc. Am.* 102, 2892-2905] toward predicting microscopic speech perception data. Model predictions were computed for the extensive consonant perception data set provided by Zaar and Dau [(2015), *J. Acoust. Soc. Am.* 138, 1253-1267], obtained with consonant-vowels (CVs) in white noise. The predictions were in good agreement with the perceptual data both in terms of consonant recognition and confusions. The model was further evaluated with respect to perceptual artifacts induced by (i) different hearing-aid signal processing strategies and (ii) simulated cochlear-implant processing, based on data from DiNino *et al.* [(2016), *J. Acoust. Soc. Am.*, 140, 4404-4418]. The model successfully predicted the strong consonant confusions measured in these conditions. Overall, the results suggest that the proposed model may provide a valuable framework for assessing acoustic transmission channels and hearing-instrument signal processing.

4:00

2pPPb9. A fast test for determining the edge frequency of a dead region. Josef Schlittenlacher (Experimental Psych., Univ. of Cambridge, Downing Site, Cambridge CB2 3EB, United Kingdom, js2251@cam.ac.uk), Richard E. Turner (Eng., Univ. of Cambridge, Cambridge, United Kingdom), and Brian C. Moore (Experimental Psych., Univ. of Cambridge, Cambridge, United Kingdom)

The presence and edge frequency, f_e , of a dead region in the cochlea can be diagnosed using psychophysical tuning curves (PTCs). When the signal frequency, f_s , falls in a dead region, the tip of the PTC lies close to f_e , rather than close to f_s . However, measurement of PTCs is time consuming, limiting their application in clinical practice. We have developed a fast test based on Bayesian active learning. Instead of estimating an entire PTC, we estimate parameters of an individual hearing model, including f_e . The task is to detect a fixed signal in the presence of a masker whose level and frequency vary across trials. After each trial, the next masker level and frequency are chosen to produce maximum reduction of the uncertainty about the parameters. The results for four participants tested so far were close to those obtained using "fast" PTCs. The Bayesian procedure has two advantages compared to PTCs: it allows quantification of the reliability of the subjects, estimated from the standard deviation of a cumulative Gaussian fitted to the psychometric function; and masker levels and frequencies can be restricted to being close to the estimated minimum of the PTC, avoiding unnecessary presentation of high level sounds.

MONDAY AFTERNOON, 26 JUNE 2017

BALLROOM A, 1:20 P.M. TO 5:20 P.M.

2pPPc

Psychological and Physiological Acoustics: Localization, Binaural Hearing, and Cocktail Party (Poster Session)

Eugene Brandewie, Chair

Research, GN ReSound, Department of Psychology, 75 E. River Rd., Minneapolis, MN 55455

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

2pPPc1. Investigating processing delay in interaural time difference discrimination by normal-hearing children. Z. Ellen Peng, Taylor Fields, and Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53711, zpeng49@wisc.edu)

Recent work suggests that interaural time difference (ITD) thresholds are adult-like by 8-10 years of age in children with normal hearing (NH). If processing time is considered, however, we hypothesize that the ability to successfully extract ITDs is not fully mature in children. A novel paradigm was designed to simultaneously measure eye gaze with an eye tracker during an ITD discrimination task with mouse-click. Stimuli were 4 kHz transposed tones modulated at 128 Hz, tested with a 3-interval, 2-alternative forced choice task (left- or right-leading ITDs) with feedback. During each trial, gaze position on the computer screen was simultaneously recorded from stimulus onset to the time when a mouse click indicated either a left or right response. Processing times were extracted from the eye gaze data and compared with those from young NH adults. This presentation will focus on

the developmental differences observed when threshold estimation is used versus when processing time is assessed via eye gaze measures. Results from this study will provide a better understanding of the developmental trajectory of binaural hearing abilities in NH children. [Work supported by NIDCD (R01DC003083 and R01DC008365 to Ruth Litovsky).]

2pPPc2. Influence of spatial and non-spatial feature continuity on cortical alpha oscillations. Golbarg Mehraei (Elec. Eng., Tech. Univ. of Denmark, Reverdilsdgate 10 3 TH, Copenhagen 1701, Denmark, gmehraei@gmail.com), Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA), and Torsten Dau (Elec. Eng., Tech. Univ. of Denmark, Lyngby, Denmark)

In everyday environments, listeners face the challenge of parsing the sound mixture reaching their ears into individual sources. Perceptual continuity of acoustic features, such as pitch or location, has an obligatory effect

on auditory attention even when a feature is not task relevant. Cortical alpha oscillations (8-12 Hz) are thought to functionally inhibit the processing of task-irrelevant information. Here, we hypothesize that discontinuities in a task-irrelevant feature disrupt the attentional modulation of alpha rhythms. Using electroencephalography in humans, we compare physiological measures during a selective auditory attention task where listeners were asked to attend to either talker, based on gender (male or female) or location (left or right). On half of the trials, a discontinuity was introduced in the task-irrelevant acoustic feature. When listeners attended to the talker, there was no evidence of alpha power lateralization, and no effect of a discontinuity in location. In contrast, when listeners attended to location, parieto-occipital alpha power increased ipsilaterally to the attended location; moreover, a discontinuity in talker reduced alpha power and disrupted alpha lateralization. Our findings support the importance of parieto-occipital alpha in suppressing sources when listeners focus spatial, but not non-spatial, attention, and show that task-irrelevant discontinuities affect these alpha rhythms.

2pPPc3. Sound source localization as a multisensory process: The Wallach azimuth illusion. M. Torben Pastore and William Yost (Speech and Hearing, Arizona State Univ., 975 South Myrtle Ave., Tempe, AZ 85287, m.torben.pastore@gmail.com)

An auditory spatial illusion, introduced by Wallach (1939, 1940) and recently revisited by Brimjoin and Ackeroyd (2012), occurs when both listeners and sounds rotate. Rotating a sound, around the listener in the azimuth plane at twice the rate of listeners' head turns, can elicit the sensation of a static sound source located either in front of the listener when the sound is originally presented from behind, or behind the listener when the sound is originally presented in front. We investigated this auditory illusion when listeners were rotated at constant velocity in a rotating chair with eyes open, for bandpass noises presented from an azimuthal ring of 24 loudspeakers. For noises that were likely to generate front-back confusions, the illusion of a stationary sound was robust, especially for low-frequency sounds. On the contrary, for noises that were unlikely to produce front-back confusions, listeners reported the sound rotating around them on the azimuth plane. These observations are predicted by a simple model, based on Wallach's original multisensory explanation, combined with estimates of the availability of spectral cues that may be used to disambiguate front/back confusions. [Partially support by a grant from the National Institute on Deafness and Other Communication Disorders, NIDCD.]

2pPPc4. Identifying a perceptually relevant estimation method of the inter-aural time delay. Areti Andreopoulou (LIMSI, CNRS, Université Paris-Saclay, LIMSI, CNRS, Université Paris-Saclay, Rue John von Neumann Campus Universitaire d'Orsay, Bât 508, Orsay 91403, France, areti.andreopoulou@gmail.com) and Brian F. Katz (Lutheries - Acoustique - Musique, Inst. ∂' Alembert, UPMC/CNRS, Paris, France)

The Inter-aural Time Difference (ITD) is a fundamental cue for human sound localization. Over the past decades, several methods have been proposed for its estimation from measured Head-Related Impulse Response (HRIR) data. Nevertheless, inter-method variations in ITD calculation have been found to exceed the known Just Noticeable Differences (JNDs), hence leading to possible perceptible artifacts in virtual binaural auditory scenes, even for cases when personalized HRIRs are being used. In the absence of an objective means for validating ITD estimations, this paper evaluates which methods lead to the most perceptually relevant results. A subjective lateralization study compared objective ITDs to perceptually driven inter-aural pure delay offsets. Results clearly indicate the first-onset *Threshold* detection method, using a low relative threshold of -30 dB, applied on 3 kHz *low-pass* filtered HRIRs as the most perceptually relevant procedure across various metrics. Alternative threshold values and methods based on the maximum or centroid of the Inter-Aural Cross Correlation of similarly filtered HRIRs or HRIR envelopes also provided reasonable results. On the contrary, phase-based methods employing the Integrated Relative Group Delay were not found to perform as well.

2pPPc5. Sound source localization identification procedures: Accuracy, precision, confusions, and misses. M. Torben Pastore and William Yost (Speech and Hearing, Arizona State Univ., 975 South Myrtle Ave., Tempe, AZ 85287, m.torben.pastore@gmail.com)

Rakerd and Hartmann (1987) provided a useful set of equations that can describe listener performance in sound source localization identification tasks requiring listeners to identify which loudspeaker presented a sound. The data from such identification tasks can be presented in confusion matrices in which one dimension is the actual sound source locations and the other dimension is the reported/perceived sound source locations. This presentation describes how Rakerd and Hartmann's measures relate to estimates of sound source localization accuracy, precision, confusions, and misses. We will describe some of the advantages and limitations of these measures of performance in sound source localization identification tasks, especially in conditions involving sound sources located around an entire azimuth circle. [Partially support by a grant from the National Institute on Deafness and Other Communication Disorders, NIDCD.]

2pPPc6. The impact of asymmetric rates on interaural time difference lateralization and auditory object formation in bilateral cochlear implant and normal hearing listeners. Tanvi D. Thakkar, Alan Kan (Waisman Ctr., Univ. of Wisconsin-Madison, 934B Eagle Heights Dr., Madison, WI 53705, thakkar@wisc.edu), and Ruth Litovsky (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Madison, WI)

Normal hearing (NH) listeners are able to accurately identify and locate sound sources using auditory object formation (AOF) and interaural time differences (ITDs). Temporal cues can further facilitate AOF and ITD sensitivity: this includes within- and across-ear stimulus' rate, envelope-, and onset-symmetry. Bilateral cochlear implant (BiCI) listeners are not guaranteed to receive symmetrical or complementing temporal information across ears. Cochlear-Nucleus devices undergo "peak-picking" where stimulation of electrodes could yield asymmetric rates across the ears, interrupting good AOF and ITD sensitivity. We investigated the impact of asymmetric rates and ITDs, on AOF and ITD lateralization. BiCI and NH listeners were presented with diotic and dichotic pulsatile stimulus rates, combined with an ITD. Rate was fixed in one ear, and varied in the contralateral ear. In a single interval, six-alternative forced-choice task, listeners reported where and how many sounds they heard. We hypothesized that with interaural asymmetries in rate, NH listeners would exhibit minimal AOF and poor lateralization, while BiCI listeners may exhibit AOF and lateralization independent of interaural rate asymmetry. The contribution of having matched rates for good AOF and ITD sensitivity helps explain the lack of successful stream segregation in BiCI listeners, whose devices do not deliver temporally-symmetric information.

2pPPc7. Vertical sound source localization when listeners and sounds rotate: The Wallach vertical illusion. M. Torben Pastore and William Yost (Speech and Hearing, Arizona State Univ., 975 South Myrtle Ave., Tempe, AZ 12180, m.torben.pastore@gmail.com)

In addition to testing his prediction that listeners would use changes in binaural cues relative to listeners' self-induced head movements to disambiguate front-back confusions, Wallach (1939) also tested his calculations that the relative rate at which these binaural cues change could be used by listeners to determine the elevation of the sound source. Wallach was able to induce the illusion that a sound source rotating along the azimuth plane was perceived as though it were above the listener. We sought to replicate and expand upon Wallach's study. We rotated listeners in a specialized chair at constant velocity. We presented filtered Gaussian noises at bandwidths of one-tenth octave, two octaves, and broadband, using center frequencies of 500 Hz and 4 kHz from a ring of 24 azimuthal loudspeakers located at pinna height. Sounds could also be presented from loudspeakers elevated relative to pinna level. The relative rates of sound and listener rotation around the azimuth plane were varied according to the relationship established by Wallach (1939) and listeners made judgments about the perceived elevation and rotation of the sounds. [Partially supported by a grant from the National Institute on Deafness and Other Communication Disorders, NIDCD.]

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2pPPc8. The roles of inhibition and adaptation for spatial hearing in difficult listening conditions. Jean-Hugues Lestang and Dan F. Goodman (Elec. and Electron. Eng., Imperial College London, 2 Philchurch Pl., London E11PG, United Kingdom, j110015@ic.ac.uk)

The computation of binaural cues such as the Interaural Time Difference (ITD) and Interaural Level Difference (ILD) by the auditory system is known to play an important role in spatial hearing. It is not yet understood how such computations are performed in realistic acoustic environments where noise and reverberations are present. It has been hypothesized that robust sound localization is achieved through the extraction of the ITD information in the rising part of amplitude modulated (AM) sounds. Dietz *et al.* (2013) tested this hypothesis using psychoacoustics and MEG experiments. They presented AM sounds with ITDs varying during the course of one AM cycle. Their results showed that participants preferentially extracted the ITD information in the rising portion of the AM cycle. We designed a computational model of the auditory pathway to investigate the neural mechanisms involved in this process. Two mechanisms were tested. The first one corresponds to the adaptation in the auditory nerve fibers. The second mechanism occurs after coincidence detection and involves a winner-take-all network of ITD sensitive neurons. Both mechanisms qualitatively accounted for the data, consequently we suggest further experiments based on similar stimuli to distinguish between the two mechanisms. Dietz *et al.* (2013), "Emphasis of spatial cues in the temporal fine structure during the rising segments of amplitude-modulated sounds," Proc. Natl. Acad. Sci. 110(37), 15151-15156.

2pPPc9. Perceived amount of reverberation consistent with binaural summation model. Gregory M. Ellis (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY 40292, g.ellis@louisville.edu) and Pavel Zahorik (Dept. of Otolaryngol. and Communicative Disord. and Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Although perceived amount of reverberation (PAR) is known to be dependent on the physical amount of reverberation present at the two ears, the extent to which this information may be combined across the ears is not well understood. Previous work using virtual auditory space techniques has demonstrated that when physical reverberation is reduced in the ear nearest a sound source while the contralateral ear is left unchanged, listeners do not report a change in PAR. Reducing physical reverberation equally in both ears, however, elicits a decrease in PAR. To better understand this phenomenon, the present study examines how PAR is affected by three additional listening conditions: scaling the contralateral ear while leaving ipsilateral fixed, scaling the contralateral ear only with no ipsilateral signal (monaural), and scaling the ipsilateral ear only with no contralateral signal (monaural). This study also examines how PAR is affected by an increase in physical reverberation present at the ears in the two listening conditions from the original study. Behavioral results are consistent with a binaural summation model that combines reverberant sound power from the two ears.

2pPPc10. Effect of frequency region on binaural interference for interaural level differences. Beth Rosen and Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, 0119E Lefrak Hall, College Park, MD 20742, brosen95@gmail.com)

Binaural interference occurs when a spectrally remote diotic interferer affects the ability to detect changes in interaural differences in the target. For interaural time differences, the magnitude of binaural interference depends on the target and interferer locations. For interaural level differences (ILDs), it is unclear if the magnitude of binaural interference depends on target and interferer location. ILDs are highly frequency dependent; ILDs become larger for increasing frequency for sources in the free field. Therefore, we hypothesized that binaural interference for ILDs would be frequency dependent, and that both target and interferer frequency would affect thresholds. In ten young normal-hearing listeners, we measured ILD discrimination thresholds using single tones, once with no interferer and then in the presence of a diotic interferer. We tested five frequencies: 500, 1000, 2000, 4000, and 8000 Hz. Each combination of target and interferer frequency was tested, resulting in 25 conditions. ILD thresholds for targets alone were frequency dependent. Binaural interference occurred for ILDs, and the magnitude changed depending on the target and interferer location.

These data will contribute to better understanding of across-frequency ILD processing, which is important for bilateral cochlear-implant users who rely on ILDs to localize sounds. [Work supported by NIH R01-DC014948 (M.J.G.)]

2pPPc11. Free-field sound localization on the horizontal plane as a function of stimulus level with an electronic, level-dependent hearing protection device. Eric R. Thompson and Zachariah N. Ennis (711th Human Performance Wing, Air Force Res. Lab, 2610 Seventh St, B441, Wright-Patterson AFB, OH 45433, eric.thompson.28@us.af.mil)

Electronic, level-dependent hearing protection devices (HPD) provide different levels of attenuation as a function of the input sound pressure level, so that loud sounds are attenuated for protection, but soft and moderate sound levels may be presented with little or no attenuation, or even with a small positive gain. While previous experiments have investigated sound localization with HPDs and moderate stimulus levels, it is important to understand the impact of level-dependent HPDs on the localization of both low- and high-level sounds. In this experiment, horizontal-plane sound localization judgments were obtained from human listeners with and without an earplug-type electronic, level-dependent HPD at several stimulus levels from 20 to 80 dB SPL. The data were analyzed in terms of the proportion of front/back reversals, and the mean absolute lateral error after correcting for front/back reversals. There were very few front/back reversals in the open-ear conditions at any level, and the mean absolute lateral error was less than 5 degrees. With the HPD, there were more front/back reversals than with open ear and the mean lateral error was greater, particularly for the loudest sounds where amplitude compression may have had an influence on performance.

2pPPc12. Source-blind localization and segregation model featuring head movement and reflection removal. Nikhil Deshpande and Jonas Braasch (Architecture, Rensselaer Polytechnic Inst., 220 3rd St., Troy, NY 12180, deshpn@rpi.edu)

This model takes two simultaneous speech signals, spatialized to unique azimuth positions and convolved with a simple multi-tap stereo impulse response. The model first identifies reflections and generates an inversion filter for the left and right channels. It then localizes the sources and virtually rotates its head to a known orientation for the best resulting segregation of the sources. Next, the model segments the input signals in time and frequency, applies the inverse filter, and searches for residual energy in each bin to compensate the target signal from the mixture. From the residual non-canceled energy, it generates a binary masking map and overlays this on the mixed signal's spectrogram to extract only the target signal. Improvement in SNR from head rotation approaches over 30 dB.

2pPPc13. Adaptation in distance perception induced by audio-visual stimuli with spatial disparity. Lubos Hladek (Inst. of Comput. Sci., P. J. Safarik Univ. in Kosice, 10-16 Alexandra Parade, New Lister Bldg. 3L, Glasgow Royal Infirmary, Glasgow G31 2ER, United Kingdom, lubos.hladek@nottingham.ac.uk), Aaron Seitz (Dept. of Psych., Univ. of California, Riverside, CA), and Norbert Kopco (Inst. of Comput. Sci., P. J. Safarik Univ. in Kosice, Kosice, Slovakia)

Simultaneous presentation of audio-visual stimuli with disparity leads to perceptual shifts in judgments of the stimulus auditory components (ventriloquism effect, VE). The shifts can persist even to auditory stimuli presented alone (ventriloquism aftereffect, VA). A previous study showed asymmetrical VE and VA for visual adaptors presented closer vs. further than the auditory components [Hladek *et al.* (2014), Visual calibration of auditory distance perception, ARO #37 Abstract PS-614]. In that study, a brief flash of light (visual adaptor) or noise burst (auditory component) were presented in front of the listener at distances 0.4-2.6 m in a small reverberant room, with visual adaptor 30% closer or further than the auditory component. Here, a new analysis of the results is presented, showing that much of the previously observed asymmetries between the two directions of shift (visual-closer vs. visual-farther) can be accounted for by referencing responses to the pre-adaptation baseline, and by scaling the data with respect to V-Aligned responses using the actual physical disparity of the auditory and

visual component. However, asymmetry still persists for the VA data and VA buildup, suggesting that different neural substrates underlie VA and VE in the distance dimension. [Work supported by APVV-0452-12 and EU H2020-MSCA-RISE-2015 grant 691229.]

2pPPc14. The effects of diffuse noise and artificial reverberation on listener weighting of interaural cues in sound localization. Tran M. Nguyen (Health and Rehabilitation Sci. Graduate Program, Western Univ., London, ON, Canada) and Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., 1201 Western Rd, Elborn College 2262, London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

The reliability of interaural time and level difference (ITD, ILD) sound location cues can be degraded by noise or reverberation. In this study, we determined how weighting of ITD and ILD varied with signal-to-noise ratio (SNR) in the presence of interaurally uncorrelated background noise and with direct-to-reverberant ratio (DRR) in the presence of artificial reverberation (generated by convolving the target signal with an interaurally uncorrelated pair of impulse responses created by multiplying Gaussian noise with a decaying exponential, $RT_{60} = 500$ ms). Wideband (0.5-16 kHz) 100-ms noise-burst targets were presented over headphones using individual head-related transfer functions. ITD and ILD were manipulated by attenuating or delaying the sound at one ear (by up to 300 μ s or 10 dB), and cue weighting was computed by comparing localization response bias to imposed cue bias. Wideband (0.5-16 kHz) and low-pass (0.5-2 kHz) noise and reverberation and SNRs and DRRs from -5 to +20 dB were used. ITD dominated in quiet, anechoic conditions. ITD was downweighted and ILD upweighted with decreasing SNR for both noises. Only downweighting of ITD and upweighting of ILD were associated with decreasing DRR in wideband and low-pass reverberation, respectively. In general, listeners increased the relative weighting of ILD in more adverse listening conditions.

2pPPc15. Preserving spatial perception in reverberant environments using direct-sound driven dynamic range compression. Henrik G. Hassager, Tobias May, Alan Wiinberg, and Torsten Dau (Hearing Systems Group, Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørstedts Plads 352, Kgs. Lyngby DK-2800, Denmark, hgha@elektro.dtu.dk)

Fast-acting hearing-aid compression typically distort the auditory cues involved in the spatial perception of sounds in rooms, due to the enhancement of low-level reverberant energy portions of the sound relative to the direct sound. The present study investigated the benefit of a novel direct-sound driven compression scheme that adaptively selects appropriate time constants to preserve the listener's spatial impression. Specifically, fast-acting compression was maintained for time-frequency (T-F) units dominated by the direct sound while the compressor was linearized via longer time constants for T-F units dominated by reverberation. This novel compression scheme was evaluated with normal-hearing listeners who indicated their perceived location and distribution of virtualized speech in the horizontal plane. The results confirmed that both independent compression at each ear and linked compression across ears resulted in more diffuse and broader, sometimes internalized, sound images as well as image splits. In contrast, the novel linked direct-sound driven compressor provided the listeners with a similar spatial perception obtained with linear processing that served as a reference. Independent direct-sound driven compression created a sense of movement of the sound between the two ears, suggesting that preserving the interaural level differences via linked compression is advantageous with the proposed direct-driven compression scheme.

2pPPc16. Effects of masker similarity on the identification and localization of vehicular sounds. Mark A. Ericson and Rachel Weatherless (Human Res., Army Res. Lab., 520 Mulberry Point Rd., Aberdeen Proving Ground, MD 21005, mark.a.ericson.civ@mail.mil)

Several experiments were conducted to determine the effects of masking sound similarity on the identification and localization of airborne and ground vehicle sounds. In some experiments, natural masking sounds were played in the presence of target vehicle sounds. In other experiments, spectral and temporal cues were artificially manipulated to determine their individual effects on identification and localization performance. The

Environment for Auditory Research of the Army Research Laboratory was used to create ambient masking sounds. In general, masking sounds that were similar to the vehicle sounds in spectral and envelope shape were more effective than dissimilar sounds in reducing auditory perception abilities of the listener. Although the complexity of most vehicle sounds enabled the listeners to correctly identify and localize the vehicle sounds in many masking conditions. The results of these experiments will be discussed in terms of the aural abilities of human subjects to identify and localize vehicle sounds in various ambient masking conditions.

2pPPc17. Interaural level difference-based model of speech localization in multi-talker environment. Peter Toth and Norbert Kopco (Inst. of Comput. Sci., Pavol Jozef Safarik Univ. in Kosice, Srobarova 2, Kosice 04154, Slovakia, peter.toth@upjs.sk)

Horizontal localization is based on extraction of the interaural time and level differences (ITD, ILD). Even in complex scenes with multiple talkers and reverberation, the auditory system is remarkably good at estimating the individual talker locations. Several previous models proposed mechanisms that stressed the importance of ITD in the localization process. Here, we examine whether azimuth estimation in complex scenarios can be based solely on ILD. We implemented a model (based on Faller and Merimaa, 2004) in which azimuth estimation was based on ILDs of signal parts with high interaural correlation and with spectral profile matching that of the target. Comparison with experimental data (Kopco *et al.*, 2010) showed that highly correlated parts of the signal, if available, provide reliable ILD estimates sufficient for precise target localization. However, for lateral target positions, at which the target dominates one ear but not the other, interaural correlation was too low to guide ILD extraction. In such cases, a new model based on finding maximum ILD provided good estimates even if maskers dominated in the worse ear. The combined model predictions matched the experimental data with target locations between -50° and 50° and for 4 maskers in reverberation. [Work supported by APVV-0452-12, H2020-MSCA-RISE-2015 #691229.]

2pPPc18. The source and effects of binaural cue ambiguity in free-field stereo sound localization—Behavioral testing. Colton Clayton, Leslie Balderas, and Yi Zhou (Speech and Hearing Sci., Arizona State Univ., 975 S. Myrtle Ave, SHS, Tempe, AZ 85287, ctclayto@asu.edu)

Horizontal sound localization in free field requires integration of interaural time (ITD) and level (ILD) differences, in making accurate spatial judgments. Recently, we showed that listeners demonstrated great variability in localizing a stereo sound source (Montagne and Zhou, JASA, 2016). We hypothesized that this variability might arise from conflicting sidedness between ITDs and ILDs within and/or across frequency bands. To test this hypothesis, here we generated a new set of stimuli with variable spatial congruence between ITDs and ILDs by adding a constant inter-channel level cue (± 5 dB) either aligned with or opposed to the inter-channel timing cue (from -1 to 1 msec). In Experiment 1 listeners responded to 15-ms broadband noise bursts. Response variability decreased when the inter-channel timing and level cues were spatially congruent and increased when they were not. In Experiment 2 listeners responded to low- and high-pass filtered noise (1.5 kHz cutoff) for the spatially incongruent stimuli only. Response variability was much reduced but the perceived source location consistently pointed to the "wrong side," favoring the level cue. Together, the new results suggest a significant weighting role for ILDs (generated from level-based stereophony) in determining the lateral position of a stereo image.

2pPPc19. The source and effects of binaural cue ambiguity in free-field stereo sound localization—Modeling simulation. Yi Zhou and Christopher Montagne (Speech and Hearing Sci., Arizona State Univ., 975 S. Myrtle Ave., Coor 3470, Tempe, AZ 85287, yizhou@asu.edu)

Our recent study (Montagne and Zhou, JASA 2016) showed that binaural localization cues—interaural time (ITD) and level (ILD) differences—were more variably distributed when stimuli were presented stereophonically instead of from single speakers. We hypothesized that variability in listeners' responses is directly related to variability in the binaural cues imposed by the stimulus. Here, we investigate the validity of this hypothesis

by examining the distribution of ITDs and ILDs using a simulated binaural neural network. The peripheral component of this model includes an updated auditory-nerve model (Zilany *et al.*, 2014) and the central component of this model incorporated binaural correlation and level difference analysis. The decision variable was made based on combined ITD and ILD distributions across frequencies. The modeled data were analyzed and interpreted with regard to results from a parallel behavioral test. The model results suggest that low-frequency ITDs are a major cue for sound source localization even they are ambiguously distributed with multiple peaks. On the other hand, the ILD cue can strongly modulate which ITD peak dominates the perceived source location. The network analysis further investigates potential neural mechanisms by examining decision-making based on an ILD-modulated ITD network vs. separate ITD and ILD networks.

2pPPc20. Binaural detection of a Gaussian noise burst target in the presence of a lead/lag masker. Jonas Braasch, M. Torben Pastore (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu), and Yi Zhou (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

When a leading stimulus is followed shortly thereafter by another similar stimulus coming from a different direction, listeners often report hearing a single auditory event at or near the location of the leading stimulus. This is called the precedence effect (PE). We measured masked detection thresholds for a noise target in the presence of a masker composed of (1) a lead/lag noise pair with the lead ITD set the same or opposite to the target, (2) a diotic masker, and (3) a dichotic pair of decorrelated noises. If the PE results in actual elimination of the lag stimulus, we would expect lower masked thresholds when the lead ITD is opposite to that of the target, as predicted by spatial release from masking. Results show that for small lead/lag delays, detection thresholds were similar to those for the diotic masker, regardless of whether the lead ITD was the same or opposite to that of the target. For longer lead/lag delays, which are unlikely to elicit the PE, thresholds approached those measured for dichotic maskers composed of two decorrelated noises. An extended EC model is used to simulate the psychophysical results. [Work supported by NSF BCS-1539276 and NSF BCS-1539376.]

2pPPc21. How accurate dereverberation should be? Effects of underestimating and overestimating binaural room impulse responses. Nirmal Kumar Srinivasan, Alexis Staudenmeier, and Kelli Clark (Dept. of Audiol., Speech-Lang. Pathol. and Deaf Studies, Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu)

The sound arriving at the listeners' ears is influenced by the binaural room impulse response (BRIR) of the listening environment. Previous research (Srinivasan *et al.*, 2017) has suggested that removing late reflections in BRIR improves speech understanding most when spatial cues are absent. However, in real world listening scenarios, it is difficult to differentiate between the effects of early reflections and late reverberation. Here, we present data from an experiment evaluating the effects of a simple static dereverberation technique on speech understanding for two reverberant environments ($T_{60} = 1$ and 2 s). The input speech signal was dereverberated by deconvolving the speech signal under three different conditions: (1) underestimating the effects of reverberation, (2) overestimating the effects of reverberation, and (3) correct estimation of the effects of reverberation. Effects of the three deconvolving techniques on identification thresholds and amount of release from masking will be discussed.

2pPPc22. Modelling the frequency dependency of binaural masking level difference and its role for binaural unmasking of speech in normal hearing and hearing impaired listeners. Christopher F. Hauth, Thomas Brand, and Birger Kollmeier (Carl von Ossietzky (CvO) Universität Oldenburg, Medizinische Physik und Cluster of Excellence Hearing4All, Oldenburg D-26129, Germany, christopher.hauth@uni-oldenburg.de)

In binaural tone-in-noise detection experiments humans can achieve substantially lower thresholds if either noise or tone have interaural time or phase differences (ITDs / IPDs) compared to diotic presentation of tone and

noise. This effect is named binaural masking level difference (BMLD) and was mainly investigated at 500 Hz. The results obtained in these experiments were used to fit binaural processing errors in the equalization-cancellation (EC) mechanism, which is an effective model of binaural processing. In this study, the frequency dependency of BMLDs is investigated for listeners with normal hearing and high frequency hearing loss. Binaural tone-in-noise detection experiments are conducted for tone frequencies of 250, 500, 750, 1000, 1500, and 2000 Hz, and ITDs of the noise up to 5 times the period of the tested frequency. Furthermore, diotic and dichotic speech reception thresholds for low pass filtered speech in noise are measured with the same listeners. The EC mechanism with processing errors derived at 500 Hz can predict the BMLDs for the remaining frequencies, except for the 250 Hz condition. Moreover, some HI listeners show a reduced BMLD in both tone-in-noise and speech intelligibility experiments, which is not covered by the EC model with normal-hearing processing errors.

2pPPc23. Effects of clinical hearing aid settings on sound localization cues. Anna C. Diedesch (Dept. of Otolaryngology/Head & Neck Surgery, Oregon Health & Sci. Univ., 7012 Sonya Dr., Nashville, Tennessee 37209, anna.c.diedesch@vanderbilt.edu), Frederick J. Gallun (Dept. of Otolaryngology/Head & Neck Surgery, Oregon Health & Sci. Univ., Portland, OR), and G. Christopher Stecker (Hearing & Speech Sci., Vanderbilt Univ., Nashville, TN)

Sound localization cues, particularly interaural level difference (ILD) cues, are known to be affected by hearing aid processing such as wide-dynamic range compression and strong directional microphones. These distorted cues may negatively impact spatial awareness and communication in complex environments, two areas of challenge for new and experienced hearing aid users. Previously, we investigated frequency specific alterations to ILD and interaural time difference (ITD) cues using linear amplification in simulated reverberant rooms and with gaged vent sizes. In reverberation, ITD became erratic and ILD reduced; minimal effects of hearing aid venting were observed (Diedesch and Stecker, *Am. Aud. Soc.* 2016). Here, we applied that approach to hearing aid settings more typically encountered by clinical patients. Phonak Audeo receiver-in-the-canal (RIC) hearing aids were programmed to typical open-fit hearing impairments using clinically normal settings for compression algorithms and directional microphones. Recordings, collected using an acoustic manikin, were compared across hearing aid coupling (standard open and closed domes) in anechoic and simulated rooms. Frequency-specific ITD and ILD were quantified across coupling, room, and hearing aid settings. [Work supported by the F. V. Hunt Postdoctoral Research Fellowship, NIH R01-DC011548 (GCS), R01-DC011828 (FJG), and the VA RR&D NCRAR.]

2pPPc24. Individual differences in cocktail party listening: The relative role of decision weights and internal noise. Robert Lutfi, Alison Tan, and Jungmee Lee (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, 1410 E. Skyline Dr., Madison, WI 53705, ralutfi@wisc.edu)

A simulated "cocktail-party" listening experiment was conducted to determine the relative role of decision weights and internal noise in accounting for the large individual differences in performance typically observed in these experiments. The listener heard over headphones interleaved sequences of random vowels and were asked to judge on each trial whether the vowels were spoken by the same AAA or different ABA talkers. The A and B vowels had nominally different F_0 and spatial position (simulated using Kemar HRTFs), but were randomly perturbed around these values on each presentation. Decision weights for each dimension, internal noise, and efficiency measures were estimated using COSS analysis [Berg (1990). *J. Acoust. Soc. Am.* 88, 149-158]. Decision weights were nonoptimal and differed across listeners, but weighting efficiency across individuals was quite similar. Individual differences in performance accuracy ranging over 40 percentage points were largely related to differences in internal noise. The results are discussed in terms of their implications for the relative role of sensory and attentional factors affecting individual performance differences in simulated cocktail party listening. [Work supported by NIDCD 5R01DC001262-24.]

2pPPc25. Rapid learning of sound schema for the cocktail party problem. Kevin Woods (Dept. of Brain and Cognit. Sci., Massachusetts Inst. of Technol., 43 Vassar st, 46-4078, Cambridge, MA 02139, kwoods@mit.edu)

Auditory scene analysis depends on knowledge of natural sound structure, but little is known about how source-specific structures might be learned and applied. We explored whether listeners internalize “schemas”—the abstract structure shared by different occurrences of the same type of sound source—during cocktail-party listening. We measured the ability to detect one of two concurrent “melodies” that did not differ in mean pitch (nor in timbre), ensuring that only the structure of these melodies over time could be used to distinguish them. Target melodies were cued by presenting them in isolation before each mixture, transposed to avoid exact repetition. The task was to determine if the cued melody was present in the subsequent mixture. Listeners performed above chance despite transposition between cue and target. Particular melodic schemas could recur across a subset of trials within a block, as well as across blocks separated by epochs in which the schema was absent. Recurrence across trials within a block facilitated target detection, and the advantage grew over the experiment despite intervening blocks, suggestive of learning. The results indicate that rapid and persistent internalization of source schemas can promote accurate perceptual organization of sound sources that recur intermittently in the auditory environment.

2pPPc26. The effect of sound intensity on lateralization with interaural time differences. Nima Alamatsaz (Biomedical Eng., New Jersey Inst. of Technol., 323 Martin Luther King Blvd., Newark, NJ 07102, nima.alamatsaz@njit.edu), Robert M. Shapley (Ctr. for Neural Sci., New York Univ., New York, NY), and Antje Ihlefeld (Biomedical Eng., New Jersey Inst. of Technol., Newark, NJ)

Previous studies examining the effect of sound intensity on ITD lateralization disagree on whether ITD lateralization changes with increasing sound level. We tested how sound intensity affects lateralization in three experiments. In all experiments, normal-hearing listeners judged the lateralization of band-limited target noise tokens (300 to 1200 Hz, 1 s duration, 10-ms

cos-squared ramp, presented with insert earphones). For each ear and target noise, sensation level (SL) was estimated using two-down one-up adaptive tracking. Each target stimulus contained an ITD of 0, 75, 150, 225, 300, or 375 μ s and was presented at 10, 25, or 40 dB SL. In experiment 1, listeners matched the ITD of a variable-ITD pointer (25 dB SL, 300-1200 Hz, 1 s duration, 10-ms cos-squared ramp) to each of the target tokens. In experiment 2, in each two-interval trial of a 2-AFC paradigm, the standard stimulus consisted of the same noise token as in experiment 1 and the signal stimulus had a randomly chosen ITD of \pm 0, 25, 50 or 75 μ s relative to the target ITD. Listeners reported whether the sound moved to the left or to the right, and thresholds were estimated at the “50%-right” point. In experiment 3, listeners indicated the perceived laterality by visually pointing on a graphical user interface. Preliminary data suggest that sound level affects lateralization, but that individual differences require testing of a greater number listeners than have historically been assessed.

2pPPc27. Influence of source location and temporal structure on spatial auditory saliency. Zuzanna M. Podwinska (School of Computing, Sci. and Eng., Univ. of Salford, The Crescent, Salford M5 4WT, United Kingdom, z.podwinska@edu.salford.ac.uk), Bruno M. Fazenda (School of Computing, Sci. and Eng., Univ. of Salford, Manchester, United Kingdom), and William J. Davies (School of Computing, Sci. and Eng., Univ. of Salford, Salford, United Kingdom)

Hitherto, not many studies have dealt with spatial auditory saliency. Auditory attention studies concerned with spatial aspects generally concentrate on top-down selective or divided attention, e.g., where subjects are asked to attend to one source at a specific location whilst being distracted with sources from different directions. The work presented here reports on experiments in which bottom-up spatial auditory attention, or saliency, has been tested. The tests were run using a fully immersive 3D audio-visual reproduction system, where interactions between auditory and visual modalities have been included. We tested how temporal structure and absolute location of sound sources around the listener influence saliency and attention.

2p MON. PM

MONDAY AFTERNOON, 26 JUNE 2017

ROOM 201, 1:15 P.M. TO 6:00 P.M.

2pSAa

Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics: Acoustic Metamaterials II

Christina J. Naify, Chair
Acoustics, Naval Research Lab, 4555 Overlook Ave. SW, Washington, DC 20375

Chair's Introduction—1:15

Invited Papers

1:20

2pSAa1. Highly directional source radiation using isotropic transformation acoustics. Andrew Norris and Xiaoshi Su (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

Recent developments in transformation acoustics (TA) have taken advantage of the isotropic nature of conformal mappings to form gradient index lens devices, such as a two-dimensional monopole-to-quadropole lens. While this TA precisely maintains the wave equation solution within the lens the radiated field is still multi-directional and not fully efficient due to impedance mismatch and non-planar

radiation. A three-fold strategy is outlined here to achieve highly directional and impedance matched devices. First, most of the rays leaving the original circular region are mapped to a single face of a polygon. Second, the center of the radiating face is impedance matched by simply scaling the size up or down. Finally, the polygon is replaced by a two-sided crescent moon mapping which optimizes the radiation across the face of higher curvature, allowing near-field focusing and quasi-planar far-field radiation. These ideas are illustrated by example simulations. Practical design methods, including water matrix and solid matrix devices, will be discussed. [Work supported through ONR MURI.]

1:40

2pSAa2. Perfect and broadband acoustic absorption in deep sub-wavelength structures for the reflection and transmission problems. Vicente Romero-García, Noé Jiménez, Vincent Pagneux, and Jean-Philippe Groby (LAUM, UMR CNRS 6613, France, Av. Messiaen, Le Mans 72085, France, virogar1@gmail.com)

The mechanisms to achieve perfect acoustic absorption by sub-wavelength structures in both reflection and transmission problems are reported. While the mechanism consists in critically coupling a single resonance independently of its nature in the reflection problem, the mechanism becomes more complicated in the transmission problem. To tackle these issues, we use asymmetric interacting resonators, whose interaction leads to the perfect absorption condition. The analyzed system consists in a panel with a periodic distribution of thin slits, the upper wall of which being loaded by Helmholtz Resonators. The propagation in the slit is highly dispersive due to the presence of the resonators, producing slow sound conditions and down-shifting the slit resonance to low frequencies. By controlling the geometry of the resonators, the visco-thermal losses are tuned to compensate the leakage of the system and fulfill the perfect absorption condition. In the case of the reflection problem, a single resonator is enough to obtain the perfect absorption. However, in the case of transmission, using an array of identical Helmholtz resonators only quasi-perfect absorption can be obtained. A possible solution is the use of double interacting resonators, one acting as reflecting wall for the previous one. This procedure can be iteratively repeated and one can design perfect and broadband acoustic absorbers based on the rainbow trapping mechanism.

Contributed Papers

2:00

2pSAa3. Redirection and splitting of sound waves by a periodic chain of thin perforated cylindrical shells. Andrii Bozhko (Univ. of North Texas, 316 Fry St. Apt. 156, Denton, TX 76201, andrii.bozhko@my.unt.edu), Jose Sanchez-Dehesa (Universidad Politecnica de Valencia, Valencia, Valencia, Spain), and Arkadii Krokhin (Univ. of North Texas, Denton, TX)

A line of perforated cylindrical shells in air is practically transparent for sound since each individual unit is a weak scatterer. However, strong scattering occurs due to the coupling to the acoustic eigenmodes of the chain. Here we develop an analytical theory of sound transmission and scattering at a linear chain of perforated shells and predict strong anomalous effect for oblique incidence. The chain eigenmodes are weakly decaying, with symmetric profile and anomalous dispersion, or with antisymmetric profile and normal dispersion, and their excitation leads to deep minima in the transmission and 90°-redirection of the external sound. At normal incidence, only the symmetric eigenmode can be excited, otherwise both modes are excited at close frequencies. Moreover, the wave which resonates with the normal-dispersion mode is redirected along the “right” direction, whereas the wave resonating with the anomalous-dispersion mode is redirected in the “wrong” direction. Thus, a periodic chain of perforated shells may serve not only as a 90°-redirecting antenna but also as a splitter of sound waves with close frequencies. For example, an acoustic signal containing two frequencies around 3 kHz can be split into monochromatic components propagating in opposite directions along the chain, if the beat frequency is ≈ 500 Hz.

2:20

2pSAa4. Nonuniform distribution of point defects in ferroelectric phononic crystal. Chandrima Chatterjee (Phys. and Astronomy, The Univ. of MS, Oxford, MS) and Igor Ostrovskii (Phys. and Astronomy, The Univ. of MS, Lewis Hall, Rm. 108, University, MS 38677, iostrov@phy.olemiss.edu)

The photoluminescence (PL) from point defects in ferroelectric phononic crystal (FPC) is investigated at room temperature. The FPC consists of periodically poled domains 0.45-mm-long each along the x-axis in the 0.5-mm-thick z-cut LiNbO₃ wafer. The spectra of PL are excited by 310 nm ultraviolet light and registered in the range of 350 to 900 nm. The PL spectra reveal different point defects including F-center, Ba, Ar, Ne, Cr, K, Fe⁺, Xe, and others. The electrically active defects such as F-center and Fe⁺ are expected to be sensitive on a local electric polarization. In a FPC, the ferroelectric neighboring domains are inversely poled and have an opposite electric polarization. The change from polarization “up” to polarization “down”

happens across so called interdomain wall. The point defect concentrations along the neighboring domains are researched by PL-scanning, consisting of taking PL-spectra from narrow zones across the domain structure along the x-axis. This scanning reveals a nonuniform distribution of defects along the FPC. The striking result is that some of the defects such as F-center and Fe⁺ have respectively narrow extrema in PL-intensity right in the interdomain wall location. Engineering application of these findings may be new non-destructive characterization method for ferroelectric phononic crystals.

2:40

2pSAa5. An improved helical-structured acoustic metamaterial design for broadband applications. Jie Zhu (Hong Kong Polytechnic Univ., Kowloon 00000, Hong Kong, jiezhu@polyu.edu.hk)

Helical structured acoustic metamaterial has been proposed to provide non-dispersive sound wave slowdown, which is an interesting topic that not only matters about fundamental explorations of slow wave physics, but also will benefit a lot of applications. Although the effect of delay acoustic signal the sound wave phase modulation is obvious, the helical structured acoustic metamaterial only provides satisfying transmission performance over narrow frequency range. In this presentation, I will introduce an improved design developed based on the original helical structured acoustic metamaterial. Such improved design can slow down acoustic wave propagation through refractive index tuning and wave-front revolution, over a much wide spectra.

3:00

2pSAa6. A discussion of macroscopic properties for acoustic metamaterials: Models and measurements. Caleb F. Sieck, Andrea Alu (Dept. of Elec. & Comput. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 1616 Guadalupe St., Austin, TX 78712, cfsieck@utexas.edu), and Michael R. Haberman (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Macroscopic material properties are useful to describe long wavelength dynamics of inhomogeneous media. Analytically, these properties are often determined by weighted field averages, which define the effective fields of a representative unit cell. The relations between these effective fields provide macroscopic properties. In addition to traditional properties (wavenumber, impedance, density, and compressibility), recent research has shown that inhomogeneous media require coupling parameters between effective volume-strain and momentum fields, known as Willis coupling or bianisotropy. However in the absence of embedded sources, metamaterial properties are

non-unique allowing for macroscopic descriptions which only include traditional properties or traditional properties and coupling parameters. Many acoustic metamaterial measurements extract macroscopic properties using reflection and transmission coefficients of finite samples. Unfortunately, this widely used technique returns properties which relate boundary fields, not effective fields as usually assumed. Even though boundary fields may well approximate effective fields for very long wavelengths, the extracted properties are at best Bloch properties of a periodic medium and in general only apply to the exact measurement setup. The aim of this talk is to provide discussion on the issues of non-uniqueness and measurements of macroscopic properties in light of the importance of physically meaningful properties.

3:20

2pSaa7. Development of a multi-material underwater anisotropic acoustic metamaterial. Peter Kerrian (Penn State Appl. Res. Lab, University Park, PA), Amanda Hanford, Robert W. Smith, Benjamin Beck, and Dean Capone (Penn State Appl. Res. Lab, Appl. Res. Lab, PO Box 30 - MS 3230D, State College, PA 16804, ald227@psu.edu)

Previous work in the open literature has described three potential ways to create an acoustic metamaterial with anisotropic mass density and isotropic bulk modulus: (1) alternating layers of homogeneous isotropic materials, (2) perforated plates, and (3) solid inclusions. The primary focus of this work will be to experimentally demonstrate the anisotropic behavior of a metamaterial comprised of a multi-solid inclusion unit cell in water. The two material design of the unit cell consists of one material more dense and one less dense than the background fluid, which results in an effective mass density tensor for the unit cell where one component is more dense and one component is less dense than the background fluid. Successful demonstration of an anisotropic metamaterial with these effective parameters is an important step in the development of structures based on transformational acoustics.

3:40–4:00 Break

4:00

2pSaa8. Acoustic wave phenomena in Willis metamaterials. Benjamin M. Goldsberry and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bgoldsberry@utexas.edu)

The design of acoustic metamaterials requires accurate modeling of the dynamics present at all relevant time and length scales. Recent work has shown that the effective dynamic properties of inhomogeneous elastic materials can result in constitutive relations that couple strain to momentum and velocity to stress, which is often referred to as Willis coupling [Willis, *Wave Motion*, **3**(1), 1-11, (1981)]. The current work will examine macro-scale acoustic propagation for waves on different time scales in a material by solving the coupled first-order equations of motion with the constitutive relations that account for Willis coupling. Specifically, second-order perturbation theory will be used to examine the classic problem of a high-frequency, low-amplitude “signal” wave superposed on a low-frequency, high-amplitude “pump” wave. Of particular interest is the slowly changing momentum bias generated by the pump wave and implications on dynamic control of signal wave propagation. Analysis and discussion will be restricted to one-dimensional wave motion.

4:20

2pSaa9. Periodic resonance effect for the design of low frequency acoustic absorbers. Thomas Dupont, Philippe Leclaire (Dr. EA1859, Univ. Bourgogne Franche Comté, BP 31 - 49 rue Mille Bourgeois, Nevers 58027, France, thomas.dupont@u-bourgogne.fr), Raymond Panneton (GAUS, Département de Génie Mécanique, Université de Sherbrooke, Sherbrooke, QC, Canada), and Olga Umnova (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom)

This presentation examines a perforated resonant material, in which the principal perforations comprise a network of periodically spaced dead-end pores. This material can show good sound absorption at low frequencies, particularly given its relatively small thickness. In a recent study, this kind

of material was modeled by an effective fluid approach which allowed low frequency approximations. At low frequency, it was shown that the periodic array of dead-end pores increases the effective compressibility without modifying the effective dynamic density. Thereby, the resonance frequency of the material is reduced in a significant way, as is the frequency of the first sound absorption peak. Moreover, a bandgap effect occurs at high frequency for the sound transmission problem. This study suggested a new concept of micro-structure for designing low-frequency resonant acoustic absorbers. A transfer matrix approach is now proposed to model and optimize such a concept. Prototypes have been made with 3D printing and tested in an acoustic tube for sound absorption and sound transmission loss. The resonant periodicity effects have been observed, and the measurements compare well with the predictions of the transfer matrix model. Finally, an optimization of the microstructure is proposed.

4:40

2pSaa10. Negative refraction experiments in soft 3D metafluids. Thomas Brunet (I2M, Université de Bordeaux, 351, cours de la libération, Bâtiment A4 - I2M/APY, Talence 33405, France, thomas.brunet@u-bordeaux.fr), Artem Kovalenko (CRPP, Université de Bordeaux, Pessac, France), Benoit Tallon (I2M, Université de Bordeaux, TALENCE, France), Olivier Mondain-Monval (CRPP, Université de Bordeaux, Pessac, France), Christophe Aristégui, and Olivier Poncelet (I2M, Université de Bordeaux, TALENCE, France)

Physics of negative refraction has been intensively studied since the 2000s. Negative refraction is usually evidenced by a Snell’s law experiment using a prism shaped negative-index metamaterial wedge. The first experimental verification of negative index of refraction was reported in 2D resonant structures at microwave frequencies [1]. A few years later, negative refraction was demonstrated in 2D and 3D resonant optical metamaterials [2]. In acoustics, the first experimental demonstration of negative refraction was reported in 3D (non resonant) phononic crystals at ultrasonic frequencies [3]. However, 3D acoustic (random) metamaterials should also offer the possibility to explore this exotic phenomenon since the first 3D locally resonant metamaterial with a negative index has been recently demonstrated [4]. In this talk, we will report on negative refraction experiments performed in these soft 3D metafluids composed of macro-porous micro-beads randomly dispersed in a water-based gel-matrix. Negative refraction will be demonstrated by the negative deflection of an ultrasonic beam outgoing from a prism shaped metafluid in a water tank experiment. [1] Shelby *et al.*, *Science* **292**, 77 (2001). [2] Valentine *et al.*, *Nature* **455**, 376 (2008). [3] Yang *et al.*, *Phys. Rev. Lett.* **93**, 024301 (2004). [4] Brunet *et al.*, *Nat. Mater.* **14**, 384 (2015).

5:00

2pSaa11. Scattering from a fluid cylinder with strain-momentum coupled constitutive relations. Michael B. Muhlestein (Signature Phys. Branch, ERDC-CRREL, 3201 Duval Rd. #928, Austin, TX 78759, mimuhle@gmail.com), Benjamin M. Goldsberry, and Michael R. Haberman (Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Many important applications of acoustics are associated with the principle of scattering. For example, biomedical ultrasound and sonar both make use of acoustic field scattering for localization, imaging, and identification of objects. While the theory of acoustic scattering from fluid and elastic materials is well established and has been validated with numerical and physical experiments, no work has been published to describe scattering from a more general class of acoustic materials known as Willis materials. Willis materials are characterized by a bulk modulus and mass density as well as a vector that couples the pressure-strain relationship with the momentum density-particle velocity relationship. The coupling vector is the result of microstructural asymmetry. We present a theoretical description of acoustic scattering of a plane wave incident upon a cylinder exhibiting weak Willis coupling using a perturbation approach. The scattered field depends upon the orientation of the Willis coupling vector and is therefore anisotropic despite the symmetry of the geometry. The analytical model is validated through comparison with a finite element-based numerical experiment.

2pSAa12. Analysis of one-dimensional wave phenomena in Willis materials. Michael B. Muhlestein (Signature Phys. Branch, ERDC-CRREL, 3201 Duval Rd. #928, Austin, TX 78759, mimuhle@gmail.com) and Michael R. Haberman (Mech. Eng., Univ. of Texas at Austin, Austin, TX)

The primary benefit of treating a complicated acoustic system as an acoustic metamaterial (AMM) is that once the effective material properties are determined, well-established mathematical analyses may be used to describe wave propagation within the system. However, many standard analyses are not well understood for the class of materials known as Willis materials. Willis materials are characterized by constitutive relations that couple both the pressure and momentum density to both the particle velocity and the volume strain. This work presents the mathematical analysis of the propagation of a velocity pulse of finite duration within a one-dimensional Willis material in the time-domain. In particular, the propagation of the pulse is described in the context of (i) an infinite Willis material, (ii) two half-spaces where one or both display Willis coupling, and (iii) a thin coupled partition in an uncoupled background. Willis coupling is shown to affect the relationship between incident and scattered waves via a convolution rather than a simple multiplication, as is the case with uncoupled media.

2pSAa13. Acoustic valley states and acoustic valley transport in sonic crystals. Jiuyang Lu (Dept. of Phys., South China Univ. of Technol., Guangzhou, Guangdong 510641, China, phjylu@scut.edu.cn)

We report the discovery of acoustic valley states in sonic crystals and the observation of the valley transport in the domain walls. The concept of valley pseudospin, labeling quantum states of energy extrema in momentum space, is attracting attention for its potential as a new type of information carrier. Inspired by the recent valley related phenomenon in election systems, the acoustic version of valley states in sonic crystals is studied and the vortex nature of such states is revealed. The extraordinary chirality of valley vortex states may provide new possibilities in sound manipulations and will be appealing to scalar acoustics since the absence of spin degree of freedom here. We further experimentally observe the topological valley transport of sound in sonic crystals. The acoustic valley transport is confined in the domain walls of the sonic crystals and behaves negligible reflection to the interface corners. The acoustic valley transport of sound, strikingly different from that in traditional sound waveguides, may serve as the basis for designing devices with unconventional functions.

MONDAY AFTERNOON, 26 JUNE 2017

ROOM 204, 1:20 P.M. TO 5:40 P.M.

2pSAb

Structural Acoustics and Vibration and ASA Committee on Standards: Novel Treatments in Vibration Damping

Kenneth Cunefare, Cochair

Georgia Tech, Mechanical Engineering, Atlanta, GA 30332-0405

Manuel Collet, Cochair

Dynamic of Complex systems, CNRS LTDS, Ecole Centrale de Lyon, 36 av G. de Collongue, Ecully 69131, France

Invited Papers

1:20

2pSAb1. Vibration damping materials with enhanced loss due to microstructural nonlinearity. Stephanie G. Konarski, Mark F. Hamilton, and Michael R. Haberman (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu)

One conventional approach to attenuating structure-borne waves or reduce the ringdown time of modes in structural elements is to attach damping layers or patches to vibrating components. Common patch and layer materials are specially formulated polymers or engineered polymeric composites that demonstrate elevated viscoelastic loss factors in the frequency range of interest. Recent research has shown that small volume fractions of negative stiffness inclusions embedded in a lossy material generate effective loss factors that exceed that of the host material. The ability to generate negative stiffness behavior, however, is often the result of nonlinear inclusion material response. Presented here is a multiscale model of a particulate composite material consisting of a nearly incompressible host material containing small-scale heterogeneities with a nonlinear elastic stress-strain response. We investigate the nonlinear dynamic behavior of the heterogeneous medium for small harmonic perturbations about several pre-strain states to demonstrate the influence of the microscale dynamic response on the macroscopically observable mechanical loss. Of primary interest is the energy dissipation capabilities of the composite which can be tuned using inclusion pre-strain. Loss for composites with nonlinear inclusions is compared to conventional composites containing air voids or steel inclusions. [Work supported by ONR.]

1:40

2pSAb2. Versatile hybrid sandwich composite combining large stiffness and high damping: spatial patterning of the viscoelastic core layer. Marta Gallo, Renaud G. Rinaldi, Laurent Chazeau, Jean-Marc Chenal (MATEIS CNRS UMR 5510 (Materials: Eng. and Sciences), Université de Lyon, INSA-Lyon, Université Lyon 1, 7, Ave. Jean Capelle, Villeurbanne 69621, France, marta.gallo@insa-lyon.fr), François Ganachaud (IMP, CNRS UMR 5223 (Polymer Mater. Eng. lab.), Université de Lyon, INSA-Lyon, Université Lyon 1, Villeurbanne, France), Quentin Leclerc, Kerem Ege, and Nicolas Totaro (LVA (Acoust. and Vibrations Lab.), Université de Lyon, INSA-Lyon, Villeurbanne, France)

With the aim of decreasing CO₂ emissions, car producers' efforts are focused, among others, on reducing the weight of vehicles yet preserving the overall vibrational comfort. To do so, new lightweight materials combining high stiffness and high (passive) damping are sought. For panels essentially loaded in bending, sandwich composites made of two external metallic stiff layers and an inner polymeric (i.e. absorbing) core are broadly used. In the present work, the performances of such sandwich structures are enhanced by optimizing their damping behavior according to their use. More precisely, spatial patterning through selective UV irradiation of the viscoelastic properties of the silicone elastomeric layer is obtained based on a recently published UV irradiation selective technique [1]. Initially developed to modulate the elastic property gradient in Liquid Silicone Rubber (LSR) membranes, the procedure is now generalized to control the viscoelastic behavior of Room Temperature Vulcanization (RTV) silicone. Since the Young's modulus and damping factor of the polymeric material are triggered by the UV irradiation dose, the resulting vibration response of the sandwich composite, made of aluminum skins and RTV silicone core, can be accordingly tuned. [1] Stricheret *et al.* (2016). "Light-Induced Bulk Architecturation of PDMS Membranes," *Macromol. Mater. Eng.* 301(10), 1151-1157.

2:00

2pSAb3. Extreme impact mitigation by critical point constraints on elastic metamaterials. Ryan L. Harne, Justin Bishop, Daniel C. Urbanek, Quanqi Dai, and Yu Song (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., E540 Scott Lab, Columbus, OH 43210, harne.3@osu.edu)

A critical transition occurs between pre- and post-buckled configurations of lightly-damped structures where the relative proportions of dissipative and elastic forces may reverse, theoretically giving rise to large effective damping properties. This paper describes computational and experimental studies that investigate such fundamental theory and principles. It is found that impact mitigation capabilities of elastic metamaterials are significantly enhanced by critical point constraints. The results of one-dimensional drop experiments reveal that constrained metamaterials reduce impact force and suppress rebound effects more dramatically than conventional damping methods, while constraints nearer to critical points magnify the advantages. When embedded into distributed structures as in conventional applications, it is found that constrained metamaterials provide superior impact mitigation capabilities than solid dampers applied at the same locations. All together, the results show that critical point constraints on elastic metamaterials provide new directions for the control and suppression of impact energy with effectiveness exceeding that achieved by solid dampers having the same bulk geometry.

2:20

2pSAb4. Evidence of multimodal vibration damping using a single piezoelectric patch with a negative capacitance shunt. Vicente Romero-García, Charlie Bricault, Charles Pézerat (LAUM, UMR CNRS 6613, France, Av. Messiaen, Le Mans 72085, France, virogar1@gmail.com), Manuel Collet (CNRS, LTDS UMR 5513, Ecully, France), Adrien Pyskir, Patrick Perrard (LTDS UMR 5513, Ecully, France), and Gaël Matten (Dept. of Appl. Mech., Institut FEMTO-ST, Besançon, France)

In this work, a piezoelectric patch shunted with a negative capacitance circuit has been used to simultaneously damp several modes of a square aluminum plate at low frequencies. The active nature of such electromechanical system leads to regions of instabilities in which the highest vibration attenuation performance appears in the softening region. Once the geometry is fixed, the system has two degrees of freedom, dominated by the electrical parameters of the circuit: the resistance and the negative capacitance. We tune both the value of the negative capacitance, in order to place the structure close to the instability in the softening region, and the resistance of the circuit in such that control the losses of the system. This work shows an optimal design to simultaneously damp several modes with non-zero electromechanical coupling factors using a single shunted patch at low frequencies. The problem is solved numerically and tested experimentally with good agreement. The results show the possibility of controlling the modal response of the system, opening prospects to improve the acoustic comfort with systems using piezoelectric shunted damping circuits with small additional mass with a high tunability by only adjusting the properties of the shunt.

2:40

2pSAb5. Design and assessment of a distributed active acoustic liner concept for application to aircraft engine noise reduction. Herve Lissek, Romain Boulandet (EPFL, EPFL STI IEL LEMA, Station 11, Lausanne 1015, Switzerland, herve.lissek@epfl.ch), Sami Karkar (Ecole Centrale de Lyon, Ecully, France), Gaël Matten (FEMTO-ST, Besançon, France), Manuel Collet (Ecole Centrale de Lyon, Ecully, France), and Morvan Ouisse (FEMTO-ST, Besançon, France)

Acoustic liners are a widespread solution to reduce turbofan noise in aircraft nacelles, due to lightweight and relatively small dimensions for integration within nacelles. Although conventional liners might be designed so as to target multiple tonal frequencies, their passive principle prevents the adaptation to varying engine speeds and therefore lowers their performance during flight, especially in the take-off and landing phases. This paper presents a novel concept of active acoustic liner based on an engineered design of microphones and loudspeakers, aiming at absorbing noise over a broad frequency bandwidth. Integration issues have been taken into account so as to fit to the targeted application to aircraft engines, yielding thickness minimization, with a view to challenging existing passive, narrow-band, liners. The sound absorption performance of the proposed active lining concept is evaluated, through commercially available finite-element software, in a configuration mimicking an aeronautical insertion-loss measurement setup, and then tested in the corresponding experimental facility in the presence of flow. The results show that such a concept is readily surpassing conventional passive liners, both in terms of insertion loss value and frequency bandwidth.

2p MON. PM

3:00

2pSAb6. Damping treatment of optical breadboards. Vyacheslav Ryaboy (Light and Motion Div., MKS, 1791 Deere Ave., Irvine, CA 92606, vyacheslav.ryaboy@newport.com)

Vibration damping of honeycomb optical tables had been subject of significant research and development efforts during several decades. These efforts resulted in tuned and broadband damping treatments, as well as active damping system that found extensive practical applications. Smaller optical platforms (breadboards) are often used for vibration-sensitive

applications due to dearth of laboratory space and increased sensitivity of small optical instruments. Relatively massive damping devices that work for large tables are not applicable to smaller (usually 5 to 10 cm thick) breadboards. This paper describes a new method for damping resonance vibration of optical breadboards with honeycomb cores. The method makes use of typical modal geometry of orthotropic laminated plates by integrating a layer of highly damped material between two surfaces making it to work in shear. Physical principle, optimization procedure, practical designs, implementation, and test results of damped breadboards are presented.

3:20–3:40 Break

3:40

2pSAb7. Efficiency of an absorber involving a nonlinear membrane driven by an electro-acoustic device. Pierre-Yvon Bryk, Sergio Bellizzi, and Renaud Côte (LMA, Aix-Marseille Univ, CNRS, Centrale Marseille, LMA, 4 impasse Nikola Tesla, Marseille 13453, France, bryk@lma.cnrs-mrs.fr)

Great attention has been recently paid to employing nonlinear energy sink (NES) as an essential nonlinear acoustic absorber rather than Helmholtz absorbers. NES are based on the principle of the “Targeted Energy Transfer” (TET) that allows to transfer the energy from a primary acoustic field to the NES. In this paper, an hybrid electro-acoustic NES (hNES) is described. It is composed of an latex membrane with one face (exterior) coupled to the acoustic field (to reduce) and the other one enclosed. The enclosure includes a feedback loop composed of a microphone and a loudspeaker that control the pressure difference at the level of the membrane. Due to the hardness behavior of the membrane in non-linear deformation, the hNES can synchronize its resonance with one of the resonances of the acoustic field greater than the linear resonance of the hNES. It allows to bring out the TET toward the hNES and thus reduce noise. The feedback loop tunes the linear resonance frequency of the hNES at low level, which is a key factor for the triggering threshold of the TET. An experimental study of the hNES will be presented including TET regime characterization and influence analysis of the feedback gain.

4:00

2pSAb8. Vibration damping using a spiral acoustic black hole. Wonju Jeon and Jae Yeon Lee (KAIST, 291 Daehak-ro, Yuseong-gu, Daejeon 34141, South Korea, wonju.jeon@kaist.ac.kr)

This study starts with a simple question: can we efficiently reduce the vibration of plates or beams using a lightweight structure that occupies a small space? As an efficient technique to damp vibration, we adopted the concept of an Acoustic Black Hole (ABH) with a simple modification of the geometry. The original shape of an ABH has a straight wedge-type profile with power-law thickness, with the reduction of vibration in beams or plates increasing as the length of the ABH increases. However, in real-world applications, there exists an upper bound of the length of an ABH due to space limitations. Therefore, in this study, the authors propose a curvilinear shaped ABH using the simple mathematical geometry of an Archimedean spiral, which allows a uniform gap distance between adjacent baselines of the spiral. In numerical simulations, the damping performance increases as the arc length of the Archimedean spiral increases regardless of the curvature of the spiral in the mid- and high-frequency ranges. Adding the damping material to an ABH can also strongly enhance the damping performance while not significantly increasing the weight. In addition, the radiated sound power of a spiral ABH is similar to that of a standard ABH.

4:20

2pSAb9. Vibroacoustics behavior of self-similarly loaded sandwich structures. Jérémie Derré and Frank Simon (DMAE, ONERA, BP 74025, 2 Ave. Edouard Belin, Toulouse CEDEX 4 31055, France, jeremie.derre@onera.fr)

The reduction of structural vibrations and acoustic radiation is increasingly challenging due to lightweight material use driven by mass reduction, particularly in the aerospace industry. Trim panels are well-known materials with high acoustic transmission and radiation. This study combines semi-analytical modeling, numerical simulations and experiments on the vibroacoustic behavior of sandwich constructions locally overloaded by a pre-fractal mass distribution. The research originality lies in the overload distribution (a self-similar pattern inspired by the Cantor set) which is directly slotted within the honeycomb core, so that the structure load-bearing capability is not altered. As the mass effect spatially localizes the vibrations, the pre-fractal pattern focuses the effect of localization on some specific frequency band gaps, which reduces the total vibrational energy and thus the acoustic radiation. Before the extension to composite plates, simulations of a homogenized beam model have been computed to form a numerical modal database. Experiments have been conducted to compare and tune the mechanical model parameters. Satisfying agreement has been obtained between simulations and experiments. Finally, acoustic radiation simulations of self-similarly loaded structures have been conducted and radiation coefficients are expected to be weaker than for classical bending beams.

4:40

2pSAb10. Effects of error in a subordinate oscillator array. John A. Sterling, Joseph F. Vignola (Mech. Eng., Catholic Univ. of America, 620 Michigan Ave., Washington, DC 20064, jsterling@gmail.com), Teresa J. Ryan, William Miller (Dept. of Eng., East Carolina Univ., Greenville, NC), and Aldo A. Glean (Mech. Eng., Catholic Univ. of America, Washington, DC)

Existing research has shown that the response of a single degree of freedom resonant system can be modified by the attachment of sets of substantially smaller resonators. Such arrays of attachments, known as subordinate oscillator arrays, can increase the apparent damping of the primary structure, and the property distributions can be selected such that the collective effects result in a response of the primary resonator that is similar to an electrical band-rejection filter. Other prior work with this system has indicated high sensitivity to disorder in the individual attachment properties such that even 0.1% variation is likely to cause undesirable effects in the overall system response. Such levels of variation well below 1% are easily attributable to typical manufacturing tolerances, environmental influences, and degradation factors. This work presents experimental results of a set of prototype subordinate oscillator arrays produced with high precision additive manufacturing techniques so as to prescribe different levels of variation.

5:00

2pSAb11. Design and closed-loop control of an electroactive polymer-based tunable Helmholtz resonator. Ahmed ABBAD (Appl. Mech., FEMTO-ST Institut/ GAUS, 2500 Boulevard de L'université, département de Génie, Sherbrooke, QC J1K2R1, Canada, ahmed.abbad@usherbrooke.ca), kanty Rabenorosoa (MN2S, FEMTO-ST Institut, Besançon, Doubs, France), Morvan Ouisse (Appl. Mech., FEMTO-ST Institut/ GAUS, Besançon, France), and Nouredine Atalla (Appl. Mech., Sherbrooke Univ., Sherbrooke, QC, Canada)

A Helmholtz resonator is a passive acoustic resonator used to control a single frequency resulting from the cavity volume and the resonator neck

size. The aim of the proposed study is to present a new concept and strategy of tunable Helmholtz resonator in order to enhance acoustic performances in lower frequencies (<500 Hz). The proposed concept consists in replacing the resonator rigid front plate by an electroactive polymer (EAP) membrane. When an electric field is applied, a change is made in the mechanical properties of the EAP membrane which induce a resonance frequency shift. A closed-loop control algorithm is developed to allow a real-time adaptability. Experimental measurements are performed on the developed prototype to determine the potential of this concept in term of acoustic absorption and Transmission Loss for low-frequency.

Invited Paper

5:20

2pSAb12. Resonance of damping. Yuri Bobrovnikskii (Theor. and Appl. Acoust., Blagonravov Mech. Eng. Res. Inst., 4, Griboedov Str., Moscow 101990, Russian Federation, yuri@imash.ac.ru)

Considered is a linear forced vibrating structure (primary structure) to which another linear passive structure (absorber) is attached at a number of points. It is shown analytically that the vibration power flow from the primary structure to the absorber reaches its absolute maximum if two conditions are met simultaneously. The first condition is well known: there must be a frequency resonance when the forcing frequency coincides with one of the eigen frequencies of the primary structure/absorber system. The second condition is also of the resonant type: at the frequency resonance vibration mode, the amount of damping in the absorber must be equal to the amount of damping in the primary structure. This can be called the resonance of damping. The vibration or sound absorber that satisfies both resonance conditions can be called the best or the perfect absorber. The theoretical result obtained has been verified in a laboratory experiment with a simple primary structure and a dynamic vibration absorber as well as in an impedance tube on a resonant sound absorber. Relation to results known from the literature and possible applications using metamaterials are discussed.

2p MON. PM

MONDAY AFTERNOON, 26 JUNE 2017

ROOM 304, 1:15 P.M. TO 5:20 P.M.

2pSC

Speech Communication: New Trends in Imaging for Speech Production

Jennell Vick, Cochair

Psychological sciences, Case Western Reserve University, 11635 Euclid Ave., Cleveland, OH 44106

Maureen Stone, Cochair

University of Maryland Dental School, 650 W. Baltimore St., Rm. 8207, Baltimore, MD 21201

Chair's Introduction—1:15

Invited Papers

1:20

2pSC1. Integrating optical coherence tomography with laryngeal videostroboscopy. Daryush Mehta (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, One Bowdoin Square, 11th Fl., Boston, MA 02114, daryush.mehta@alum.mit.edu), Gopi Maguluri, Jesung Park (Physical Sci., Inc., Andover, MA), James B. Kobler (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, Boston, MA), Ernest Chang, and Nicusor Iftimia (Physical Sci., Inc., Andover, MA)

During clinical voice assessment, laryngologists and speech-language pathologists rely heavily on laryngeal endoscopy with videostroboscopy to evaluate pathology and dysfunction of the vocal folds. The cost effectiveness, ease of use, and synchronized audio and visual feedback provided by videostroboscopic assessment serve to maintain its predominant clinical role in laryngeal imaging.

However, significant drawbacks include only two-dimensional spatial imaging and the lack of subsurface morphological information. A novel endoscope will be presented that integrates optical coherence tomography that is spatially and temporally co-registered with laryngeal videoendoscopic technology through a common path probe. Optical coherence tomography is a non-contact, micron-resolution imaging technology that acts as a visual ultrasound that employs a scanning laser to measure reflectance properties at air-tissue and tissue-tissue boundaries. Results obtained from excised larynx experiments demonstrate enhanced visualization of three-dimensional vocal fold tissue kinematics and subsurface morphological changes during phonation. Real-time, calibrated three-dimensional imaging of the mucosal wave and subsurface layered microstructure of vocal fold tissue is expected to benefit in-office evaluation of benign and malignant tissue lesions. Future work calls for the *in vivo* evaluation of the technology in patients before and after surgical management of these types of lesions.

1:40

2pSC2. New trends in visualizing speech production. Jamie Perry (East Carolina Univ., College of Allied Health Sci., Greenville, NC 27858, perryja@ecu.edu)

Most dynamic magnetic resonance imaging of the levator veli palatini (levator) muscle has been limited to studies using sustained phonation (Yamawaki *et al.* 1996; Ettema *et al.* 2002; Tian *et al.*, 2010a, 2010b; Kollara and Perry, 2013). Dynamic MRI sequences at faster imaging speeds using word-level productions have been described along the midsagittal image plane (Sutton *et al.*, 2009; Bae *et al.*, 2011; Scott *et al.*, 2013) and oblique coronal image plane (Perry *et al.*, 2013b, 2013c; Scott *et al.*, 2013). The purpose of this presentation is to describe methods for analyzing the velopharyngeal mechanism and velar muscles during rest and speech production using MRI. This presentation will demonstrate a potentially useful technique in dynamic MRI that does not rely on cyclic repetitions or sustained phonation and can provide dynamic information related to muscle function. Innovative techniques in imaging and interpreting dynamic MRI of velopharyngeal function during speech will be described. It is expected these developments will provide insights that will improve clinical methods in resonance assessments among children born with cleft lip and palate.

2:00

2pSC3. Three dimensional MRI analyses of tongue muscle behavior. Maureen Stone (Univ. of Maryland Dental School, 650 W. Baltimore St., Rm. 8207, Baltimore, MD 21201, mstone@umaryland.edu), Jiachen Zhuo, Nahla ElSaid (Radiology, Univ. of Maryland Med. School, Baltimore, MD), Jonghye Woo, Fangxu Xing (Massachusetts General Hospital, Harvard, Boston, Maine), and Jerry Prince (Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD)

3D and 4D MRI provides unique, high dimensional data for use in clinical applications and scientific models. Our group is currently using diffusion tensor MRI (DTI) and tagged-cine MRI (tMRI) to explore the fiber direction of muscles in the tongue and the shortening patterns of those fibers during speech. tMRI uses “magnetic tags” to mark and track tissue points, so that when the tissue deforms, the tags reflect these deformations. Soft tissue motion patterns, such as those within the tongue during speech, provide a link between muscle activation and tongue surface shape. Tissue points also demark the endpoints and internal points of muscles, and can identify tongue muscle position and shortening during speech. Tracking tongue muscle motion is unique to tMRI because their interdigitated fibers make them fairly opaque to EMG. Disambiguating muscles and identifying their shortening pattern is a first step to relating muscle action to tongue deformation. The second tool, DTI, identifies the location and orientation of “muscle fibers” within a muscle volume. We will use these tools to track shortening of several tongue muscles during speech.

2:20

2pSC4. Multimodal investigation of speech production featuring real-time three-dimensional ultrasound. Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu)

Real-time three-dimensional ultrasound offers new opportunities to investigate the spatial and temporal complexities of speech production, but ultrasound by itself continues to be limited in significant ways because (typically) only one surface of the vocal tract can be imaged: the tongue. Nevertheless, many of these limitations can be overcome by appropriately time-aligning and spatially registering ultrasound volumetric images with additional data streams, such as lateral webcam video, acoustic recordings, and digitized palate impressions. One method by which this may be accomplished makes use of a new open-source toolbox for MATLAB, called “WASL,” which was originally designed specifically for real-time three-dimensional ultrasound, but is now extensible to other biomedical imaging modalities, including MRI, CT, and two-dimensional ultrasound. Examples from a study of rhotic sound production illustrate how multimodal investigations of speech production featuring real-time three-dimensional ultrasound may be carried out using WASL.

2:40

2pSC5. Recent results in silent speech interfaces. Bruce Denby (Tianjin Univ., Institut Langevin, 1 rue Cuvier, Paris 75005, France, bruce.denby@gmail.com), Shicheng Chen, Yifeng Zheng (Tianjin Univ., Tianjin, China), Kele Xu (Université Pierre et Marie Curie, Paris, France), Yin Yang (Univ. of New Mexico, Albuquerque, NM), Clemence Leboulenger (Université Pierre et Marie Curie, Paris, France), and Pierre Roussel (ESPCI, Paris, France)

Silent Speech Interfaces (SSI) are sensor-based communication systems in which a speaker articulates normally but does not activate their vocal chords, creating a natural user interface that does not disturb the ambient audio environment or compromise private content, and may also be used in noisy environments where a clean audio signal is not available. The SSI field was launched 2010 in a special issue of Speech Communications, where systems based on ultrasound tongue imaging, electromyography, and electromagnetic articulography were proposed. Today, although ultrasound-based SSIs can achieve Word Error Rate scores rivaling those of acoustic speech recognition, they have yet to reach the marketplace due to performance stability problems. In recent years, numerous approaches have been proposed to address this issue, including better acquisition hardware; improved tongue contour tracking; Deep Learning analysis; and the association of ultrasound data with a real time 3D model of the tongue. After outlining the history and basics of SSIs, the talk will present a summary of recent advances aimed at bringing SSIs out of the laboratory and into real world applications.

3:00

2pSC6. Discovering functional units of the human tongue during speech from cine- and tagged-MRI. Jonghye Woo (MGH/Harvard, 55 Fruit St., White 427, Boston, MA 02114, jwoo@mgh.harvard.edu), Maureen Stone (Univ. of Maryland, Baltimore, Baltimore, MD), Fangxu Xing (MGH/Harvard, Baltimore, MD), Jordan Green (MGH Inst. of Health Professions, Boston, MA), Arnold Gomez (Johns Hopkins Univ., Baltimore, MD), Van Wedeen (MGH/Harvard, Boston, MA), Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD), and Georges El Fakhri (MGH/Harvard, Boston, MA)

Tongue motion during speech or other lingual behaviors involves synergies of locally deforming regions, or functional units. Determining functional units will provide an insight into the mechanisms of normal and pathological muscle coordination, leading to improvement in understanding of speech production and treatment or rehabilitation procedures. In this work, we present an approach to determining functional units using cine- and tagged-MRI. Functional units are estimated using a sparse non-negative matrix factorization (NMF) framework, learning latent building blocks and the weighting map from motion features from displacements and strain. Current models of gesture production suggest that, during speech planning, talkers select temporal frames prior to specifying the appropriate spatially fixed clusters. Our analysis is intended to parallel this process by using NMF to first identify temporal frames in the data based on change points in motion features, and then to identify the spatially fixed clusters for all the input quantities in each time frame. A spectral clustering is performed on the weighting map of each time interval to define the coherent sub-motions, revealing temporally varying tongue synergies. Synthetic and human tongue data including both controls and patients with glossectomy and amyotrophic lateral sclerosis are used to define subject/task-specific functional units of the tongue in localized regions.

3:20–3:40 Break

3:40

2pSC7. Seeing is treating: 3D electromagnetic midsagittal articulography (EMA) visual biofeedback for the remediation of residual speech errors. Jennell Vick (Psychol. Sci., Case Western Reserve Univ., 11635 Euclid Ave., Cleveland, OH 44106, jennell@case.edu), Rebecca Mental (Psychol. Sci., Case Western Reserve Univ., Cleveland Heights, OH), Holle Carey (Vulintus, Dallas, TX), and Gregory S. Lee (Elec. Eng. and Comput. Sci., Case Western Reserve Univ., Cleveland, OH)

Production distortions or errors on the sounds /s/ and /t/ are among the most resistant to remediation with traditional speech therapies, even after years of weekly treatment sessions (e.g., Gibbon *et al.*, 1996; McAuliffe & Cornwell, 2008; McLeod, Roberts, & Sita, 2006). In this study, we report on the results of treating residual speech errors in older children and adults with a new visual biofeedback treatment called Opti-Speech. Opti-Speech uses streaming positional data from the Wave EMA device to animate real-time motion of a tongue avatar on a screen. Both the clinician and the client can visualize movements as they occur relative to target shapes, set by the clinician, intended to guide the client to produce distortion-free and accurate speech sounds. Analyses of positional data and associated kinematics were completed during baseline, treatment, and follow-up phases for four participants, two who produced pre-treatment residual errors on /s/, and two with residual errors on /t/. Measures included absolute position of the 5 tongue sensors, variability of position, perceptual quality ratings, and acoustic measures of the consonants. Results indicate that Opti-speech effectively remediated the residual speech errors with corresponding evidence of generalization to untreated contexts and maintenance of improvements at follow-up.

4:00

2pSC8. Computer simulation of the vocal tract in speech production. Ian Stavness, Erik Widing, Francois Roewer-Despres (Comput. Sci., Univ. of SK, 110 Sci. Pl., Saskatoon, SK S7N5C9, Canada, ian.stavness@usask.ca), and Bryan Gick (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Neuro-musculoskeletal function can be well characterized for most human movements, such as walking, through a combination of experimental measurements, including motion capture, electromyography, and force sensing. Movements of the vocal tract, however, pose a considerable challenge in terms of holistically measuring and visualizing the neuro-musculoskeletal processes underlying speech production. This is due to the inaccessibility of the vocal tract, the large number of small, deep muscles involved, the fast speed of movements, and the inherent three-dimensional nature of vocal tract shape during speech production. An array of measurement and imaging modalities have been tailored to vocal tract measurement, such as MRI, EMA, EPG, and ultrasound, each providing valuable, but incomplete information regarding vocal tract movements. Computer simulation of the vocal tract can play an important role in complementing these sparse experimental measurements by aiding in the fusion of multiple imaging modalities, helping to describe and visualize the 3D structures of interest, filling in the gaps in experimental measurements (both spatial and temporal), and extrapolating from the small sample sizes commonly found in speech production studies. We will discuss two types of computer simulations that are emerging as important complementary forms of speech production investigation: forward biomechanical simulation of the vocal tract articulators and probabilistic simulation of neuro-musculoskeletal parameters.

Contributed Papers

2p MON. PM

4:20

2pSC9. Maintaining vowel distinctiveness in children with Steinert Myotonic Dystrophy: An ultrasound study. Marie Bellavance-Courtemanche, Pamela Trudeau-Fisette, Amélie Prémont, Christine Turgeon (Linguist, Université du PQ a Montreal, CP 8888, succ. Centre-Ville, Montreal, QC H3C 3P8, Canada), and Lucie Menard (Linguist, Université du PQ a Montreal, Montreal, QC H3C 3P8, Canada, menard.lucie@uqam.ca)

Speech production entails appropriately timed contractions of many muscles. Steinert myotonic dystrophy, a neurodegenerative disease that causes muscle weakness and difficulties in muscle relaxation after muscle contraction, frequently affects orofacial articulatory dynamics leading to decreased speech intelligibility. We aimed to investigate the articulatory and acoustic characteristics of cardinal vowels produced by children with Steinert disease. We recruited fourteen 6- to 14-year-old French-speaking children diagnosed with Steinert disease and 14 aged-matched typically developing children. They were asked to produce repetitions of the vowels /i a u/ in consonant-vowel (CV) contexts. A synchronized ultrasound, Optotrak motion tracking system, and audio recording system was used to track lip and jaw displacement as well as tongue shape and position. Duration and formant values were also extracted. The Euclidean distance between vowels, in the formant space, was reduced in children with Steinert disease compared to the control children. Different patterns of articulatory contrasts were observed among the children, with some of them using more tongue contrasts than lip contrasts. Intelligibility tests conducted with adult listeners on a subset of the data show that some patterns are related to higher intelligibility scores than others.

4:40

2pSC10. Predicting velum movement from acoustic features—A regression approach. Hong Zhang (Linguist, Univ. of Pennsylvania, 619 Williams Hall, 255 S 36th St., Philadelphia, PA 19104, zhangho@sas.upenn.edu)

Although acoustical methods have been widely used in the nasality literature, a direct link between the acoustical measurements and velum movement during speech production is yet to be established. In this study, we propose a model through which the vertical movements of the velum are inferred from an acoustic feature set. An X-ray Microbeam data set collected at University of Tokyo are used for the modeling. The data recorded the vertical movements of the velum of 11 American English speakers saying both isolated words and sentences. Velum positions are recorded from

tracing a metal pallet placed on top of the velum. 40 MFCC (Mel-frequency Cepstral Coefficient) features are extracted from the accompanying acoustic signal at each time frame. MFCCs of ten frames before the current frame, together with the current frame, consist of the feature vector for predicting the velum movement of the current frame. Elasticnet regression is used to reduce the dimensionality of the feature vector. In general, MFCCs from higher frequencies are penalized during model selection. The selected features are then fitted to a stepwise logistic model. For each individual speaker, the inferred velum movements in the validation set are a good fit to the actual observation as judged by the high accuracy in identifying locations of peaks and valleys and small deviance from the response. However, there exists large inter-speaker variation in terms of both movement pattern and model performance.

5:00

2pSC11. Measuring regional displacements of tongue parts on ultrasound during /ɹ/ articulation. Sarah M. Hamilton, Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, 3433 Clifton Ave., Cincinnati, OH 45220, hamilsm@mail.uc.edu), Neeraja Mahalingam, Allison Garbo (Biomedical, Chemical, and Environ. Eng., Univ. of Cincinnati, Cincinnati, OH), Ashley Walton, Michael A. Riley (Psych., Univ. of Cincinnati, Cincinnati, OH), and T. Doug Mast (Biomedical, Chemical, and Environ. Eng., Univ. of Cincinnati, Cincinnati, OH)

The ability to differentiate movements of the front, back and root portions of the tongue is important to the development of mature speech coordination. This ability is particularly relevant for sounds with complex tongue shapes, such as the American English rhotic approximant /ɹ/ (“r”), but speakers also have a wider scope of coarticulatory opportunities if able to control tongue parts independently [Zharkova, 2012]. In addition, lack of independence in tongue part movement is associated with speech sound disorders [Gibbon, 1999; Gick *et al.*, 2008]. For this study, relative displacements of tongue blade, dorsum, and root were analyzed using MATLAB-based image processing. Regions of interest (ROIs) were drawn for these three areas on ultrasound images during production of /ɑɹ/ by 25 adults. Displacements of each region were measured by tracking of local brightness maxima from images representing /ɑ/ and /ɹ/ production, resulting in ranges of relative blade, dorsum and root displacement associated with normal /ɑɹ/ production. These ranges are compared to data from a sample of ultrasound images of speech from children with persistent speech errors. Preliminary results suggest that children with persistent sound errors have a smaller range of tongue part displacement when compared to typical adults.

2pSP

Signal Processing in Acoustics, Engineering Acoustics, and Architectural Acoustics: Signal Processing for Directional Sensors II

Kainam T. Wong, Chair

Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, DE 605, Hung Hom KLN, Hong Kong

Invited Papers

1:20

2pSP1. Circular arrays of directive sensors for improved direction-of-arrival estimation. Houcem Gazzah (Elec. and Comput. Eng., Univ. of Sharjah, M9-223, University City, Sharjah, Sharjah 27272, United Arab Emirates, hgazzah@sharjah.ac.ae)

In directing finding, it is highly desired to design antenna arrays with constant accuracy across the whole field-of-view. The so-called array isotropy is achieved, most notably, by means of the uniform circular array. On the other hand, sensors with directive response improve performance but unequally, even if displayed in circular configurations. It follows that arbitrarily oriented arrays can suffer severe loss in performance. We show (i) what conditions are needed in order to preserve array isotropy while using directive sensors, and (ii) what can be done to compensate for the loss of isotropy when this condition is not met. For instance, we show that there is a critical array size below which the Cramer Rao Bound, our performance measure, is not direction-independent. In some cases, the direction-dependent expression can be simplified to the point where a closed-form array-adaptation algorithm is obtained. It calculates the optimal array orientation as function of the available statistical information about the source. Depending on how much is known about the source, the amount of improvement may be very significant compared to cases where the array is arbitrarily oriented, up to 50% for arrays of cardioid sensors.

1:40

2pSP2. A biomimetic coupled circuit based microphone array inspired by the fly *Ormia ochracea*. Xiangyuan Xu, Ming Bao, and Han Jia (Key Lab. of Noise and Vib. Res., Inst. of Acoust., Chinese Acad. of Sci., No. 21, West Rd., Beisi Huan, Haiding District, Beijing City 100190, China, xiangyuxu699@gmail.com)

Miniaturization of microphone arrays poses challenges to attain accurate localization performance due to shrinking aperture. The fly *Ormia ochracea* is able to determine the direction of sound source with an astonishing degree of precision, even though its ears are extremely small. Inspired by this, an equivalent analog circuit is designed to mimic the coupled ear system of the fly for sound source localization. This coupled circuit receives two signals with tiny phase difference from a space closed two-microphone array, and produces two signals with obvious intensity difference. The response sensitivity can be adjusted through the coupled circuit parameters. The directional characteristics of the coupled circuit have been demonstrated in the experiment. The designed miniature microphone array can localize the sound source with low computational burden by using the intensity difference. This system has significant advantages in various applications where the array size is limited.

Contributed Paper

2:00

2pSP3. LWD-DATC: Logging-While-Drilling Dipole Shear Anisotropy estimation from Two-Component waveform rotation. Pu Wang (Mitsubishi Elec. Res. Lab., Cambridge, MA), Sandip Bose, Bikash K. Sinha, Ting Lei (Mathematics Modeling, Schlumberger-Doll Res., Cambridge, MA), and Matthew Blyth (Schlumberger, 300 Schlumberger Dr., Sugar Land, TX 77478, MBlyth@exchange.slb.com)

Sonic dipole shear anisotropy orientation in subsurface formations is key information for a complete characterization of rock mechanical models for well planning. In *Wireline* logging, it is estimated with the well-known Alford rotation on four-component inline and crossline waveforms from two orthogonal dipole firings. In contrast, shear orientation estimation is more challenging in *Logging-While-Drilling* (LWD) operations because of the

following reasons:—No exactly orthogonal cross-dipole firings in LWD tools due to tool rotation while drilling;—Coupling between the *formation* flexural mode and the strong *collar* flexural mode propagating through the stiff drill collar;—Unavoidable tool eccentricity;—Strong drilling noise from vibration, shock, and the turbulent mud flow around the tool; To overcome the aforementioned challenges, this paper describes a new technique to estimate the LWD dipole shear orientation with two-component waveforms obtained from a single dipole firing. It is accomplished by maximizing the projected energy of the two-component waveforms into a subspace defined by two eigenfunctions (the Bessel functions of the first- and second-kinds) accounting for the propagation of the two coupled flexural modes over multiple frequency points. By resorting to the subspace estimation, the new technique has been successfully validated on synthetic data and tested on field data sets.

Invited Paper

2:20

2pSP4. Far field directivity of an omnidirectional parametric loudspeaker consisting of piezoelectric ultrasonic transducers set on a sphere. Oriol Guasch, Patricia Sánchez-Martín, and Marc Arnela (GTM-Grup de recerca en Tecnologies Mèdia, Dept. of Eng., La Salle, Universitat Ramon Llull, C/ Quatre Camins 30, Barcelona, Catalonia 08022, Spain, oguasch@salleurl.edu)

In this work, it is proposed to extend the convolution model for the far-field directivity of a parametric loudspeaker array (PLA) in [C. Shi and Y. Kajikawa, *J. Acoust. Soc. Am.* 137 (2) (2015)] to predict the far-field sound field generated by an omnidirectional parametric loudspeaker (OPL). The original two-dimensional model, intended for flat PLAs, relies on convolving the directivity of the primary waves with the Westervelt one, rather than on performing their product. This allows one to deal with piezoelectric ultrasonic transducers (PZTs) having large beam widths, typical of PLA applications. The model is herein enhanced to three dimensions, to predict the far-field pressure level of the difference wave at any observation point in space, for a PZT located and pointing anywhere. This makes the model amenable to compute the far-field directivity of PLAs on curved surfaces. In particular, it is shown how it can be applied to compute the far-field pressure of an OPL consisting of a spherical surface with hundreds of PZTs placed on it.

Contributed Papers

2:40

2pSP5. Passive detection of low frequency sources using vector sensor channel cross correlation. Thomas J. Deal (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, thomas.deal@navy.mil)

Acoustic vector sensors consisting of an omnidirectional hydrophone and three orthogonal velocity sensors offer new opportunities for passive detection of acoustic sources at low frequencies. In the frequency range from 10 to 300 Hz, the predominant noise sources in the ocean are surface wave action and distant shipping. Omnidirectional hydrophones are affected by both noise types whereas vertically-oriented velocity sensors are more affected by surface wave action, and horizontally-oriented velocity sensors

are more affected by distant shipping. Under these ambient conditions, when no source is present, the vector sensor channels have low cross-correlation levels. When a source is present, propagation conditions cause an increase in correlation between all vector channels. This increase is measurable in the magnitude and phase of the correlation coefficients between each channel pair, which forms the basis of an algorithm for detecting acoustic sources. This paper presents the response of an acoustic vector sensor to both types of ambient noise and demonstrates the change in correlation when a source is present. These noise and signal responses determine a detection threshold for a single frequency source for desired probabilities of detection and false alarm.

3:00–3:20 Break

3:20

2pSP6. Directive and focused acoustic wave radiation by tessellated transducers with folded curvatures. Ryan L. Harné, Danielle T. Lynd, Chengzhe Zou, and Joseph Crump (Mech. and Aeronaut. Eng., The Ohio State Univ., 201 W 19th Ave., E540 Scott Lab, Columbus, OH 43210, harné.3@osu.edu)

Guiding radiated acoustic energy from transducer arrays to arbitrary points in space requires fine control over contributions of sound provided to the point from each transducer constituent. Recent research has revealed advantages of mechanically reconfiguring array constituents along the folding patterns of an origami-inspired tessellation, in comparison to digitally processing signals sent to each element in a fixed configuration. To date, this concept of acoustic beamfolding has exemplified that far field wave radiation may be adapted by orders of magnitude to a point according to the folding of a Miura-ori tessellated array when the array constituents are driven by the same signal. This research investigates a new level of adaptive acoustic energy delivery from foldable arrays through study of tessellated transducers that adopt folded curvatures, thus introducing opportunity for

near field energy focusing alongside far field directionality. The outcomes of these computational and experimental efforts plainly reveal that foldable, tessellated transducers that curve upon folding empower straightforward means for the fine, real-time control needed to beam and focus sound to points in space. Discussions are provided on the potentials and limitations of the current tessellations considered, and means for future studies to build upon the new understanding.

3:40

2pSP7. Cramer-Rao bound for direction finding at a tri-axial velocity-sensor of an acoustic event having an AR(1) temporal auto-correlation. Dominic M. Kitavi, Kainam T. Wong (Dept. of Electron. and Information Eng., Hong Kong Polytechnic Univ., Hong Kong Polytechnic University, Yuk Choi Rd., Kowloon, HungHom 0852, Hong Kong, dominic.kitavi@connect.polyu.hk), Lina YEH (Dept. of Mathematics, Soochow Univ., Taipei, Taiwan), and Tsair-Chuan Lin (Dept. of Statistics, National Taipei Univ., New Taipei City, Taiwan)

Various acoustical events have an order-1 autoregressive temporal auto-correlation model. This investigation derives the Cramer-Rao bound (CRB) for direction finding of such an AR(1) acoustical event, if the data is collected by a tri-axial velocity-sensor.

Invited Paper

2pSP8. An array of two biaxial velocity-sensors of non-identical orientation— Their “spatial matched filter” beam-pattern’s pointing error. Chibuzo J. Nnonyelu (Electron. and Information Eng., The Hong Kong Polytechnic Univ., BC606 (Table 3), Hong Kong Polytechnic University, 11 Yuk Choi Rd., Hung Hom, Hung Hom, Kowloon 999903, Hong Kong, joseph.nnonyelu@connect.polyu.hk), Charles H. Lee (Dept. of Mathematics, California State Univ., Fullerton, Fullerton, CA), and Kainam T. Wong (Electron. and Information Eng., The Hong Kong Polytechnic Univ., Hong Kong, Kowloon, Hong Kong)

A biaxial velocity-sensor (a.k.a. a v-v probe) measures two Cartesian components of an incident acoustical wavefield’s particle velocity vector. Such biaxial velocity-sensors are sometimes used as elements in an array. If these elements are not identically oriented, what would happen to the array’s “spatial matched filter” beam-pattern? This paper investigates such a beam-pattern’s pointing error, for a pair of non-identically oriented biaxial velocity-sensor. This pointing error is investigated here in terms of (i) the biaxial velocity-sensor’s misorientation skew angle, (ii) the biaxial velocity-sensor’s spatial separation, (iii) the incident signal’s wavelength, and (iv) the beamformer’s nominal look direction.

Contributed Papers

4:20

2pSP9. Higher-order estimation of acoustic intensity. Joseph S. Lawrence, Kent L. Gee, Tracianne B. Neilsen, and Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, joseph-lawrence@hotmail.com)

The phase and amplitude gradient estimator (PAGE) method [D. C. Thomas *et al.*, *J. Acoust. Soc. Am.* **137**, 3366-3376 (2015)] can be used to increase the bandwidth of complex acoustic intensity estimates obtained with multi-microphone probes. Despite the increased bandwidth, errors arise when using this method, which is based on linear least-squares gradients, in non-planar fields. Examples of non-planar fields include the acoustic near field of a radiating source or near a null in a standing-wave field. The PAGE method can be improved by using a Taylor expansion to obtain higher-order estimates of center pressure, pressure amplitude gradient, and phase gradient. With a sufficient number of microphones, the higher-order method is shown to improve the bandwidth of both the active and reactive intensity estimates. Additionally, this method can be used to estimate the spatial dependence of intensity across the extent of the probe. [Work supported by the National Science Foundation.]

4:40

2pSP10. Direction finding using a “p-v probe” of higher order. Muhammad Muaz (Dept. of Electron. and Information Eng., The Hong Kong Polytechnic Univ., CD-515, Hong Kong 999077, Hong Kong, m.muaz@connect.polyu.hk), Yue I. Wu (College of Comput. Sci., Sichuan Univ., Chengdu, China), Da Su, and Kainam T. Wong (Dept. of Electron. and Information Eng., The Hong Kong Polytechnic Univ., Hong Kong, Kowloon, Hong Kong)

The “p-v probe” comprises one isotropic pressure-sensor and one uni-axial velocity-sensor. The “p-v probe” has been used for acoustic direction finding for a few decades. This work generalizes the “p-v probe,” by allowing a higher-order “figure-8” directional sensor to substitute for the uni-axial velocity-sensor, which represents a first-order “figure-8” directional sensor. This work presents new eigen-based azimuth-elevation direction-finding algorithms in closed forms, as well as the corresponding Cramèr-Rao lower bounds.

5:00

2pSP11. A distributed network of compact microphone arrays for drone detection and tracking. Aro Ramamonjy (Acoust. and Protection of the Combattant, French-German Res. Inst. of Saint-Louis, LMSSC, 2 rue Conté, Paris 75003, France, aroramamonjy@gmail.com), Eric Bavu, Alexandre Garcia (Cnam, Laboratoire de Mécanique des Structures et des Systèmes Couplés (EA3196), Conservatoire National des Arts et Métiers, Paris, France), and Sébastien Hengy (Acoust. and Protection of the Combattant, French-German Res. Inst. of Saint-Louis, Saint-Louis, France)

This work focuses on the development of a distributed network of compact microphone arrays for unmanned aerial vehicle (UAV) detection and

tracking. Each compact microphone array extends in a 10 cm length aperture and consists in an arrangement of digital MEMS microphones. Several arrays are connected to a computing substation using the I2S, ADAT and MADI protocols using optical fiber. These protocols used together allow to collect the signals from hundreds of microphones spread over a distance of up to 10 km. Sound source localization is performed on each array using measured pressure and particle velocities. The pressure is estimated by averaging the signals of multiple microphones, and the particle velocity is estimated with high order finite differences of microphone signals. Multiple calibration procedures are compared experimentally. Results in sound source localization, noise reduction by spatial filtering and UAV recognition using machine learning are presented.

5:20

2pSP12. Dual accelerometer vector sensor mounted on an autonomous underwater vehicle (AUV)—Experimental results. Paulo J. Santos, Paulo Felisberto, Frederich Zabel, Sergio Jesus (University of Algarve, LARSyS, Campus da Penha, Faro, Faro 8005-139, Portugal, pjsantos@ualg.pt), and Luís Sebastião (IST/ISR, University of Lisbon, LARSyS, Lisbon, Lisbon, Portugal)

In seismic geo-acoustic exploration, the use of ships equipped with long streamers is of major concern due to the complexity of the operations. The European project WIMUST aims to improve the efficacy of actual geo-acoustic surveys through the use of AUVs towing short streamers. A Dual Accelerometer Vector Sensor (DAVS) was developed in order to complement the streamer’s data, allowing for the reduction of their size and facilitating the operation of the WiMUST distributed configuration. Each DAVS consists of two tri-axial accelerometers and one hydrophone aligned along a vertical axis. This configuration has the ability to cancel or significantly attenuate the direct and the surface reflection paths, which are undesirable in seismic imaging. Calibration tests with the DAVS have already been performed; this paper presents experimental results on the estimation of azimuthal directions when the DAVS is in motion. Signals in the 1-2kHz band were emitted by a source deployed in a shallow pond at 1.5m depth and acquired by the DAVS mounted on a MEDUSA class AUV, which was following a pre-programed path with a 0.26m/s nominal speed. The azimuth estimates are coherent with the MEDUSA trajectories even in curved paths where the thruster noise increases.

2pUWa**Underwater Acoustics and Acoustical Oceanography: In Honor of Ira Dyer, 60 Years as an Innovator, Entrepreneur, and Visionary for Ocean Engineering**

Arthur B. Baggeroer, Cochair

Mechanical and Electrical Engineering, Massachusetts Inst. of Technology, Room 5-206, Cambridge, MA 02139

Peter Mikhalesky, Cochair

Leidos, 4001 N. Fairfax St., Arlington, VA 22207

Philip Abbot, Cochair

*OASIS, Inc., 5 Militia Drive, Lexington, MA 02421***Chair's Introduction—1:15*****Invited Papers*****1:20****2pUWa1. Ira Dyer and the BBN Applied Research Division.** James Barger (Raytheon BBN Technologies, Raytheon BBN Technologies, 10 Moulton St., Cambridge, MA 02138, docobra1@aol.com)

Ira Dyer was one of the first hires by Leo Beranek for the fledgling company Bolt, Beranek and Newman (BBN). Ira in turn hired many of the now recognized pioneers in ocean, structural; and ocean acoustics and formed the BBN Applied Research Division. Members investigated all aspects of sound and vibration in ships, submarines, aircraft, and spacecraft. An example includes during the mid-1950s, Ira helped design the US Navy X-1 Submarine, a small four-man quiet diesel-electric submarine for ADM Rickover. BBN designed an innovative engine mounting system that quieted the vehicle and led the way for ultra-quiet submarines in the future Navy. This submarine is now on display at the Navy Submarine Museum at Groton, CT. Other contributions by members include SEA (statistical energy analysis) and criteria for modeling flow-excitation forcing functions to structural radiation. Ira created a golden era for acoustics at BBN.

1:40**2pUWa2. The contribution of Ira Dyer to understanding hydro-structural acoustics.** William Blake (Mech. Eng., Adjunct Professor, The Johns Hopkins Univ., 6905 Hillmead Rd., Bethesda, MD 20817, hydroacoustics@aol.com)

Ira Dyer's work in structural acoustics is well known, but what I think is little known is his start in aeroacoustics followed by his early work in the response of structures to pressure fields of a kind that is typical of hydroacoustics. Lighthill's seminal paper on aerodynamic noise in 1952 is generally regarded as the dawn of modern flow acoustics. Immediately after this, in 1953 and continuing into the early 1960s, Ira began his career by examining scaling laws for the sound and the structural vibration associated with jets and high speed flow over plate and shell structures. In this work he examined scaling laws for sound power, the importance of wave coincidence in fluid-structure interaction, and the effects of space-time decorrelation of excitation pressure fields. This paper will examine the conclusions of these early papers of Ira's and trace their relevance through to modern times and across the combined fields of hydro-structural acoustics. Ira's professional path led him away from these areas of interest but his early work nonetheless laid foundations for work of the future.

2:00**2pUWa3. Ira Dyer at MIT—Professor, Department Head, and Arctic pioneer.** Arthur B. Baggeroer (Mech. and Elec. Eng., Massachusetts Inst. of Technol., Rm. 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu) and Peter Mikhalesky (Leidos, Arlington, VA)

Ira arrived at MIT in 1970 from BBN to take a professorship offered by Alfred Kyle, Head of the Department of Naval Architecture. He had said yes; however, Kyle surprised him and asked him to also head the department, renamed Ocean Engineering. With some trepidation he accepted. He started new research programs in ocean acoustics, ambient noise, reverberation and propagation while making seminal contributions. He became the director of the MIT Sea Grant Program and soon MIT was one of the first Sea Grant colleges. In 1977 Ira had the idea to research basin-scale reverberation. He thought the Mediterranean would be the ideal enclosed basin including opportunities for post-experiment R&R! He approached the Office of Naval Research, they enthusiastically agreed, and sent him to the Arctic! That detour north turned into a super highway of decades of Arctic acoustics research from a controversial seamount discovery

in his first reverberation experiment, detailed morphology of ice cracking noise, propagation and ice scattering to global climate change. Ira will have lasting and enduring impact in acoustics through his contributions and through his many students and colleagues that had the great privilege and joy to learn from, know, and work with Ira.

2:20

2pUWa4. Ira Dyer and MIT Arctic acoustics research. Gregory L. Duckworth (Raytheon BBN Technologies, 10 Moulton St., Cambridge, MA 02138, gregory.duckworth@raytheon.com)

Ira Dyer and his colleague Art Baggeroer began MIT's Arctic Acoustics research efforts in the latter half of the 1970s. In this pre-GPS and newly digital era, Ira helped define an ambitious program for the Office of Naval Research that brought large-aperture high-quality acoustic array data acquisition to bear on the problems of Arctic acoustic science. Supporting a generation of graduate students, this work explored ambient noise generation mechanisms in the ice, Arctic basin reverberation, seismic reflection and refraction studies to understand the underlying crustal structure, and long-range propagation that is now exploited for acoustic thermometry to study global climate change. Advanced signal processing, combined with fundamental physical understanding and modeling of the ice, water column, and underlying crust vastly increased our understanding of these phenomena in the Arctic. I will show some of the results of this work, and how it has been extended for use in active sonars and measurements of sea-ice dynamics over large areas of the Arctic.

2:40

2pUWa5. The role of fluctuations in the interpretation of sonar detection performance. Philip Abbot, Vincent E. Premus, Mark N. Helfrick, Charles J. Gedney, and Chris J. Emerson (OASIS, Inc., 5 Militia Dr., Lexington, MA 02421, abbot@oasislex.com)

The predictive probability of detection (PPD) is a metric that accounts for uncertainty in sonar detection performance due to random fluctuations in transmission loss, noise level, and source level [P. Abbot and I. Dyer, *Impact of Littoral Environmental Variability on Acoustic Predictions and Sonar Performance*, 2002]. It is well known that a significant portion of Ira's career was dedicated to understanding the role fluctuations play in interpreting acoustic measurements. Building on this foundation, we now embrace the notion that a useful statement of sonar system performance is one linked with a probabilistic description of the acoustic environment's intrinsic variability. In this paper, we discuss the impact of fluctuations on passive sonar performance, including how PPD facilitates the interpretation of sonar system recognition differential. Data from a recent field test conducted in August, 2011, on the New Jersey continental shelf, will be used to illustrate the methodology and interpret measured detection performance in the presence of a cold pool duct and variable ambient noise conditions.

3:00–3:20 Break

3:20

2pUWa6. Acoustic intensity fluctuation studies in shallow water over the past quarter century. James Lynch (Woods Hole Oceanographic, MS # 11, Bigelow 203, Woods Hole Oceanographic, Woods Hole, MA 02543, jlynch@whoi.edu), John A. Colosi (Naval Postgrad. School, Monterey, CA), Timothy F. Duda, Ying-Tsong Lin, and Arthur Newhall (Woods Hole Oceanographic, Woods Hole, MA)

One of Ira Dyer's seminal contributions to underwater acoustics was in the understanding of acoustic intensity fluctuations. In particular, the 5.6 dB intensity fluctuation of a large number of interfering narrowband multipaths, which is often informally referred to as "the Dyer number", is a robust and well known effect. In shallow water, a number of physical effects can modify this basic number. These effects have been measured in various experiments over the years, and we will show a subset of these results. The physics of the effects, the measurement techniques, and the implications of this work for sonar processing will all be discussed. Work supported by ONR.

3:40

2pUWa7. Analysis of marginal ice zone noise events: Revisited. Chi-Fang Chen (Eng. Sci. and Ocean Eng., National Taiwan Univ., No. 1 Roosevelt Rd. Sec.#4, Taipei 106, Taiwan, chifang@ntu.edu.tw)

Acoustic transients observed in a noise time series, called noise events, are believed to be the major contributor to Arctic Ocean ambient noise. Underwater ambient noise data from the Marginal Ice Zone Experiment 1984 (MIZEX84), at frequencies between 20 and 2000 Hz, are studied. Results show that temporal and spatial distribution of events are characterized by clustering and seem to depend on environmental conditions such as swells and ice concentration. Event densities increase as f at frequencies below 200 Hz, decrease as f^{-1} at frequencies above 300 Hz, and reach a maximum at frequencies between 200 and 300 Hz. The change in trend may indicate a transition in the source mechanism for frequencies below 200 Hz and above 300 Hz. The noise source can be modeled as octo-pole which is the best fit from the noise data. [Work supported by Office of Naval Research, and completed under Prof. Ira Dyer's supervision in Massachusetts Institute of Technology in 1990.]

4:00

2pUWa8. Hybrid solution techniques for structural acoustics. Charles Corrado (Appl. Physical Sci. Corp., 475 Bridge St., Ste. 100, Groton, CT 06340, ccorrado@aphysci.com), Matthew Conti, Ann Stokes (Appl. Physical Sci. Corp., Lexington, MA), and Edward Heyd (Appl. Physical Sci. Corp., Littleton, CO)

Ira Dyer's structural acoustics research at MIT emphasized developing an understanding of wave interaction mechanisms and the role of structural complexity in dictating energy transport, dissipation, and radiation to the surrounding media. His work stressed a combination of experimentation and analytic-numeric methods to delineate the essential wave propagation mechanisms governing the response of the structure and the radiated field. Here we present an extension of research Ira led to study acoustic scattering from internally loaded shell structures. Hybrid analysis techniques employing wave transmission methods to represent plate and shell structures,

and analytic or finite element methods to represent attached structures provide a means to discern the partitioning between wave types comprising the response. The method directly represents all allowable wave types and quantifies their displacement functions and dispersion behavior enabling detailed evaluation of energy transport mechanisms. Interaction loads produced at attached structures are computed using shell structure Green's function matrices, impedance matrix descriptions of the attachments, and unloaded shell response vectors. The method is illustrated for marine pile driving and aerospace liftoff applications. Broadband impulse response maps are shown to illustrate the temporal evolution of the field over the structure and complement frequency domain evaluations providing further physical insight.

4:20

2pUWa9. Ira Dyer advisor to the Navy. Henry Cox (Lockheed Martin, 4350 N. Fairfax Dr., Ste. 470, Arlington, VA 22203, harry.cox@lmco.com)

Among his many activities, Ira Dyer was a valued consultant to the Navy. He was called to participate as a charter member of the Submarine Superiority Technical Advisory Group established by Admiral De Mars in 1995. He served on this and other panels for more than 15 years. He made many important contributions to the submarine force, particularly relating to ocean acoustics pertaining to operations, and structural acoustics pertaining to submarine design. Ira's ability to penetrate difficult problems, simplify and explain was unique. When Ira spoke, every one listened. Comments on his contributions and recollections of his colleagues are presented.

4:40

2pUWa10. An Office of Naval Research perspective on Ira Dyer's Legacy. Robert H. Headrick (Code 32, Office of Naval Res., 875 North Randolph St, Arlington, VA 22203, bob.headrick@navy.mil)

The talk will highlight a few of the many important contributions Professor Dyer made to underwater acoustics through his interactions with ONR as a researcher, advisor, and educator. His wide ranging expertise included sound and vibration in complex structures that led the way for the U.S. Navy to develop ultra-quiet submarines, as well as the statistics of sound propagation in the ocean, submarine sonar design, and advances in sonar signal processing. Beginning in 1978, he led and participated in six Arctic field programs, including the Canadian Basin Arctic Reverberation Experiment that imaged the entire Arctic basin with acoustics. Over the course of these efforts, that he really enjoyed talking about, he and his students developed a systematic understanding of sound propagation and noise in the Arctic.

5:00

2pUWa11. The vision of Ira Dyer. Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., 5-212, Cambridge, MA 02139, makris@mit.edu)

We will describe how Ira Dyer's scientific ideas and vision have been passed on to us and survive as strong as ever influencing the future.

Contributed Paper

5:20

2pUWa12. Changing Arctic ambient noise. Andrew J. Poulsen (Appl. Physical Sci., 49 Waltham St., Lexington, MA 02421, poulsen@alum.mit.edu) and Henrik Schmidt (Massachusetts Inst. of Technol., Cambridge, MA)

The Arctic Ocean is undergoing dramatic changes, with the most apparent being the rapidly reducing extent and thickness of the summer ice cover. Furthermore, a persistent inflow of a shallow tongue of warm Pacific water has recently been discovered in the Beaufort Sea region of the Arctic, often called the Beaufort Lens, which creates a strong acoustic duct between

approximately 100 and 200 m depth. These changes have had a significant effect on underwater acoustic propagation and noise properties. In spring 1994, acoustic data was collected in the Beaufort Sea region of the Arctic using a suspended vertical array; in spring 2016, similar data was collected in the same region. The 1994 data features meandering narrow-band features due to ice ridge friction, while the 2016 data in the new Arctic is largely dominated by ice mechanical events at discrete ranges and bearings. Supported by acoustic noise modeling, we illustrate these and other noise properties measured more than two decades apart in a region of rapid and significant change. [Work supported by ONR and DARPA.]

2pUWb**Underwater Acoustics: Sound Propagation and Scattering in Three-Dimensional Environments II**

Ying-Tsong Lin, Cochair

Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543

Frédéric Sturm, Cochair

*Acoustics, LMFA, Centre Acoustique, Ecole Centrale de Lyon, 36, avenue Guy de Collongue, Ecully 69134, France****Invited Papers***

1:20

2pUWb1. Three-dimensional model benchmarking for cross-slope wedge propagation. Orlando C. Rodríguez (LARSyS, Campus de Gambelas - Universidade do Algarve, Faro, N/A PT-8005-139, Portugal, orodrig@ualg.pt), Frédéric Sturm (Ctr. National de la Recherche Scientifique, Ecully, France), Pavel S. Petrov (V.I. Il'ichev Pacific Oceanological Inst., Vladivostok, Russian Federation), and Michael Porter (HLS Res. Inc., La Jolla, CA)

Cross-slope wedge propagation is considered for the three-dimensional benchmarking of three underwater acoustic models, one based on normal mode theory and the other two based on ray tracing. To this end, the benchmarking relies on analytic solutions for adiabatic and non-adiabatic propagation, as well as on experimental data from a scale tank experiment (a previous benchmarking with a parabolic equation model is known to provide extremely accurate predictions in all cases). The benchmarking allows to identify the advantages, accuracy and limitations of the considered models.

1:40

2pUWb2. Examining the intra-modal interference in an idealized oceanic wedge using scale-model experiments and acoustic propagation modeling. Jason D. Sagers and Megan S. Ballard (Environ. Sci. Lab., Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu)

Scale-model tank experiments are beneficial because they offer a controlled environment in which to make underwater acoustic propagation measurements that can provide high-quality data for comparison with numerical models. This talk presents results from a 1:7500 scale model experiment for a wedge with a 10° slope fabricated from closed-cell polyurethane foam to investigate three-dimensional (3D) propagation effects. A $333 \mu\text{s}$ second long pulse allows the acoustic field to obtain a steady-state, continuous-wave signal. A computer controlled positioning system accurately moves the receiving hydrophone in 3D space to create a dense field of vertical line arrays, which are used to mode filter the measured time series. The single-mode fields show the classical interference pattern resulting from rays launched up and along the slope. The measured data are compared to an exact, closed-form solution for a point source in wedge with impenetrable boundaries. The finite size of the source and the departure from the perfectly reflecting boundary conditions are discussed. The measured data are also compared to results from a three-dimensional ray model known as Bellhop3D which can account for the non-ideal boundary condition. [Work supported by ONR.]

2:00

2pUWb3. Measurements and modeling of three-dimensional acoustic propagation in a scale-model canyon. Megan S. Ballard and Jason D. Sagers (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

Scale-model acoustic propagation experiments were conducted in a laboratory tank to investigate three-dimensional (3D) propagation effects induced by range-dependent bathymetry. The model bathymetry, patterned after measured bathymetric data, represents a portion of the Hudson Canyon at 1:7500 scale and was fabricated from closed-cell polyurethane foam using a computer-numerically controlled (CNC) milling machine. In the measurement apparatus, a computer-controlled positioning system precisely locates the receiving hydrophone which permits the creation of synthetic horizontal line arrays. Results are shown for propagation paths along and across the axis of the canyon. The measurements are explained using both a hybrid method known as vertical modes/horizontal parabolic equation and a 3D ray model known as Bellhop3D. For propagation along the canyon axis, horizontal focusing is observed and discussed in the context of normal modes. For the across canyon propagation, the reflective foam walls of the canyon scatter sound back toward the receiver array from out of plane. This effect is examined using the 3D ray trace. For both cases, the capabilities of the models and their computation details are discussed. [Work supported by ONR.]

2pUWb4. Decoherence effects in 3D fluctuating environments: Numerical and experiment study. Gaultier Real (DGA Techniques Navales, Ave. de la tour royale, Toulon 83100, France, gaultier.real@gmail.com), Xavier Cristol (Thales Underwater Systems, Sophia-Antipolis, France), Dominique Habault (Laboratoire de Mécanique et d'Acoustique, Aix-Marseille Université, CNRS, Centrale Marseille, Marseille, France), and Dominique Fattaccioli (DGA Techniques Navales, Toulon, France)

This paper is devoted to the study of the effects of ocean fluctuations on acoustic propagation. The development of an ultrasonic test-bench allowing to reproduce, under laboratory conditions, the influence of 3D fluctuations on received acoustic data is presented. The experimental protocol consists in transmitting, in a water tank, a high-frequency wavetrain throughout an acoustic slab presenting a plane input face and a randomly rough output face. The various regimes of saturation and unsaturation classically used in the literature are explored by tuning the statistics of the so-called *RAFAL* (*R*AndoM *F*aced *A*coustic *L*ens). Comparisons to a “corresponding” oceanic medium are obtained via a scaling procedure. In parallel, numerical tools were developed to provide meaningful comparison with the acquired data. Both based on a split-step Fourier algorithm, a 3D PE simulation of the tank experiment and a 3D PE simulation of real scale acoustic propagation programs are presented. Features of acoustic fields perturbed by internal waves are found. The relevance of our procedure is evaluated through calculations of the coherence function (in particular, measurement of the radius of coherence) and statistical distributions of the received complex pressure and intensity. Comparisons between our scaled measurements, numerical computations and analytical results are analyzed.

Contributed Papers

2:40

2pUWb5. Experimental modal decomposition of acoustic field in cavitation tunnel with square duct test section. Romuald Boucheron (DGA HydroDynam., Chaussée du Vexin, Val-de-Reuil 27105, France, romuald.boucheron@intradef.gouv.fr), Sylvain Amailland, Jean-Hugh Thomas, Charles Pézerat (LAUM, Le Mans, France), Didier Fréchou, and Laurence Briançon-Marjollet (DGA HydroDynam., Val-de-Reuil, France)

The operational requirements for naval and research vessels have seen an increasing demand for quieter ships either to comply the ship operational requirements or to minimize the influence of shipping noise on marine life. To estimate the future radiated noise of a ship, scale measurements are realized in cavitation tunnel. DGA Hydrodynamics operates its cavitation tunnel with low background noise which allows such measurements. Understanding acoustic propagation in cavitation tunnel remains a challenge. The success of an accurate acoustic measurement depends both on a realistic propagation model and also on an efficient control of acoustic sensor characteristics. This short communication presents the results of experiments performed in GTH (Large Cavitation Tunnel) at DGA Hydrodynamics. An acoustic source radiates pure sine wave at the entrance of the test section and generates an acoustic field measured with flush mounted hydrophones. A modal decomposition is then performed to fit measurements. Complex amplitudes of all propagative modes could be estimated both for upstream and downstream propagation. Results show that for a given frequency range, modal decomposition could be an accurate model for acoustic propagation. Furthermore, different configurations of the test section and source locations have been investigated and show acoustic properties of the tunnel.

3:00

2pUWb6. Vertical underwater acoustic tomography in an experimental basin. Guangming Li, David Ingram (Inst. of Energy System, Univ. of Edinburgh, Faraday Bldg., King's Buildings, Colin Maclaurin Rd., Edinburgh, Scotland EH9 3DW, United Kingdom, G.Li@ed.ac.uk), Arata Kaneko, Minmo Chen (Graduate School of Eng., Hiroshima Univ., Higashi-Hiroshima, Hiroshima, Japan), Noriaki Gohda (Graduate School of Eng., Hiroshima Univ., Higashihiroshima, Japan), and Nick Polydorides (Inst. of Digital Communications, Univ. of Edinburgh, Edinburgh, United Kingdom)

Ocean acoustic tomography is well developed for monitoring environmental parameter changing for mesoscale ocean distribution. Small scale underwater acoustic tomography could be used for flow profile mapping in experimental tank. Vertical acoustical tomography analyse is a key aspect for 3D mapping of flow current profile. This article investigates vertical underwater acoustic tomography in a circular multidirectional wave/current basin. Two modified coastal acoustic tomography (CAT) systems were deployed in the 25 m diameter circular basin. High frequency (50 kHz) M-sequence signal was transmitted reciprocally for time of flight. The 2 m depth water column was divided into 5 layers for layered analysis. Multipath arrivals reflected by surface and bottom were distinguished by ray tracing. The 0.8 m/s straight flow was generated along the sound propagation path. Vertical layered current velocity was analyzed by solving inverse problem. The study suggest that vertical acoustic tomography could be used in the lab test tank for flow current velocity measurement.

3:20–3:40 Break

Invited Papers

3:40

2pUWb7. Radial and azimuthal acoustic propagation effects of the continental slope and sand dunes in the northeastern South China Sea. Chi-Fang Chen (Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan), Linus Chiu (Inst. of Appl. Marine Phys. and Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan), and Ching-Sang Chiu (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyder Rd., Rm. 328, Monterey, CA 93943-5193, chiu@nps.edu)

A sound source, transmitting a 1.5-2.0 kHz chirp signal periodically, was towed at a depth of 50 m along a circular track that has a radius of 3 km centered on a vertical hydrophone array moored on the upper slope of the northeastern South China Sea during the Sand Dunes Acoustic Propagation Experiment in 2014. The largest amplitude of these sand dunes was close to 20 m with horizontal length scales between 200 and 400 m. Two-dimensional (2-D) and three-dimensional (3-D) underwater acoustic propagation models, namely FOR3D with the Nx2D option and FOR3D with the fully 3D option, were employed to simulate the acoustic propagation over the continental slope, with and without the sand dunes, from the towed source to the vertical hydrophone array. Environmental inputs to the

models were measured bathymetry and sound speed profiles, obtained from multibeam echo sounding surveys and moored oceanographic sensors, respectively. Simulation results pertaining to the 2-D and 3-D propagation effects in relation to the slope, the sand dunes and the water-column variability are presented and discussed. Simulation results are also compared to the measured transmission data. [The research is jointly sponsored by the Taiwan MOST and the US ONR.]

4:00

2pUWb8. Variability of the sound field interference pattern due to horizontal refraction in shallow water. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, Mt. Carmel, Haifa 31905, Israel, bkatsnls@univ.haifa.ac.il)

Variability of the ocean waveguide's parameters in horizontal plane (bathymetry, sound speed profile, etc.) can lead to some set of effects in the sound propagation, which are specified as horizontal refraction, or in more general form, as 3D effects. These effects are being studied both in shallow and deep water, some of them were registered in experiments, or were analyzed using different theoretical models. In given paper main attention is drawn to the theory and experimental manifestation of spatial and temporal variations of interference pattern formed by narrow-band signals. Theoretical analysis is carried out using approach "Vertical Modes and Horizontal Rays" and "Vertical Modes and PE in horizontal plane." It is shown, that dependence of the sound field distribution in horizontal plane (both in a ray or PE approaches) on frequency and mode number, combined with possible multipath propagation, may lead to rather specific observable effects: variations of the amplitude and phase fronts, evolution of signal's spectrum and shape, appearance of whispering gallery modes in vicinity of curvilinear coastal line. Results of modeling and experiments (mainly Shallow Water 2006) are presented. Data processing techniques to register 3D effects are discussed. [Work was supported by ISF.]

Contributed Papers

4:20

2pUWb9. Characteristics of bottom-diffracted surface-reflected arrivals in ocean acoustics. Ralph A. Stephen, S. T. Bolmer (Woods Hole Oceanographic Inst., 360 Woods Hole Rd., Woods Hole, MA 02543-1592, rstephen@whoi.edu), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

Bottom-diffracted surface-reflected (BDSR) arrivals are a ubiquitous feature in long-range ocean acoustic propagation. BDSRs are distinct from bottom-reflected surface-reflected (BRSR) arrivals because the angle of emergence is not equal to the angle of incidence. They are not predicted by existing forward models based on available bathymetric and bottom properties data. Research cruises in the Philippine Sea and North Pacific, in 2011 and 2013, respectively, were carried out to understand BDSRs in more detail for transmissions out to 50 km range. Transmissions from a controlled source at about 60m depth were received on ocean bottom seismometers and a deep vertical line array of hydrophones. In the North Pacific experiment alone over 40 distinct bottom diffractor locations were identified. Based on these data sets BDSRs can be characterized in terms of: (a) in-plane or out-of-plane diffractors, (b) diffractor location relative to bathymetric features, (c) grazing angle of the incident field, (d) transmission frequency (from 75 to 310 Hz), (e) receiver type (vertical or horizontal seismometer, hydrophone, etc.), (f) receiver location, and (g) signal strength relative to direct and BRSR paths. [Work supported by ONR.]

4:40

2pUWb10. Multi-modal and short-range transmission loss in ice-covered, near-shore Arctic waters. Miles B. Penhale and Andrew R. Barnard (Mech. Eng. - Eng. Mechanics, Michigan Technol. Univ., 1400 Townsend Dr., R.L. Smith MEEM Bldg., Houghton, MI 49931, mbpenhal@mtu.edu)

Prior to the 1970s, extensive research has been done regarding the sound propagation in thick (kilometers) ice sheets in Arctic and Antarctic environments. Due to changing climate conditions in these environments, new experimentation is warranted to determine sound propagation characteristics in, through, and under thin-ice sheets (meters). In April 2016, several experiments were conducted approximately 1 mile off the coast of Barrow, Alaska on shore-fast, first year ice, approximately 1.5 m thick. To determine the propagation characteristics of various sound sources, Frequency Response Functions (FRFs) were measured between a source location and several receiver locations at various distances from 50 m to 1 km. The primary sources used for this experiment were, an underwater speaker with various tonal outputs, an instrumented impact-hammer on the ice, and a propane cannon that produced an acoustic blast wave in air. In addition, several anthropogenic sources, namely, a snowmobile, generator, and ice auger, were characterized. The transmission characteristics of the multipath propagation (air, ice, and water) are investigated and reported.

5:00

2pUWb11. Sound propagation in deep water with an uneven bathymetry. Zhiguo Hu, Zhenglin Li, Renhe Zhang, Yun Ren (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21, North Fourth Ring Rd. West, Haidian District, Beijing 100190, China, hzhg@mail.ioa.ac.cn)

Water depth has significant effects on sound propagation in underwater. A deep-water propagation experiment along two different tracks with the flat and uneven bottoms was conducted in the South China Sea in 2014. Some different propagation phenomena and horizontal-longitudinal correlations oscillation patterns were observed. Due to the reflection-blockage effects by a sea hill with height less than 1/10 water depth, transmission losses increase up to about 8 dB in the reflection area of the sea hill than that of flat bottom. Moreover, there is an inverted-triangle shadow zone with a maximal depth of 1500 m below the sea surface. The horizontal-longitudinal correlations in the reflection area of the sea hill do not show an obvious cyclical oscillation any longer as that in flat bottom environment. The differences of the transmission losses and the horizontal-longitudinal correlations oscillation patterns are explained by using the ray theory. [Work supported by the National Natural Science Foundation of China under Grant Nos. 11434012, 41561144006, and 11404366.]

5:20

2pUWb12. Acoustic propagation from the transitional area to deep water. Jixing Qin, Renhe Zhang, and Zhenglin Li (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, qjx@mail.ioa.ac.cn)

Sound propagation over the continental shelf and slope is complicated and is also an important problem. Motivated by a phenomenon in an experiment conducted in the Northwestern Pacific indicating that the energy of the received signal around the sound channel axis is much greater than that at other depths, we study sound propagation from the transitional area to deep water. Numerical simulations with different source depths are first performed, from which we reach the following conclusions. When the source is located near the sea surface, sound wave will be strongly attenuated by bottom losses in a range-independent environment, whereas it can propagate to a very long range because of the continental slope. When the source is mounted on the slope bottom in shallow water, acoustic energy will be trapped near the sound channel axis, and it converges more evidently than the case where the source is located near the surface. Then, simulations with different source ranges are performed. By comparing the relative energy level in the vertical direction between the numerical results and the experimental data, the range of the unknown air-gun source is approximated. The phenomenon can be confirmed by the experiment with a deterministic source located in the transitional area. [Work supported by the National Natural Science Foundation of China under Grant Nos. 11434012 and 41561144006.]

2p MON. PM

MONDAY AFTERNOON, 26 JUNE 2017

ROOM 104, 2:00 P.M. TO 3:00 P.M.

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

D. S. Houser, Vice Chair ASC S3/SC 1

National Marine Mammal Foundation, 2240 Shelter Island Drive Suite 200, San Diego, CA 92106

K. Fristrup, Vice Chair ASC S3/SC 1

National Park Service, Natural Sounds Program, 1201 Oakridge Dr., Suite 100, Fort Collins, CO 80525

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

People interested in attending the meeting of the TAGs for ISO/TC 43/SC 1 Noise and ISO/TC 43/SC 3, Underwater acoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Monday, 26 June 2017.

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

MONDAY AFTERNOON, 26 JUNE 2017

ROOM 104, 3:15 P.M. TO 4:30 P.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

C. J. Struck, Chair ASC S3

CJS Labs, 57 States Street, San Francisco, CA 94114 1401

P. B. Nelson, Vice Chair ASC S3

Department of SLHS, University of Minnesota, 115 Shevlin, 164 Pillsbury Drive S.E., Minneapolis, MN 55455

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

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Monday, 26 June 2017.

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

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

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

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

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

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Monday, 26 June 2017.

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

2p MON. PM

OPEN MEETINGS OF TECHNICAL COMMITTEES

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

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

Committees meeting on Monday, 26 June

Committee	Start Time	Room
Acoustical Oceanography	8:00 p.m.	310
Animal Bioacoustics	8:00 p.m.	313
Architectural Acoustics	8:00 p.m.	207
Engineering Acoustics	8:00 p.m.	204
Physical Acoustics	8:00 p.m.	210
Psychological and Physiological Acoustics	8:00 p.m.	311
Structural Acoustics and Vibration	8:00 p.m.	312

Committees meeting on Wednesday, 28 June

Committee	Start Time	Room
Biomedical Acoustics	8:00 p.m.	312
Musical Acoustics	8:00 p.m.	200
Noise	8:00 p.m.	203
Signal Processing in Acoustics	8:00 p.m.	302
Speech Communication	8:00 p.m.	304
Underwater Acoustics	8:00 p.m.	310