Session 3aID

Interdisciplinary: Plenary Lecture: Objective Evaluation of Musical Instrument Quality: A Grand Challenge in Musical Acoustics

Thomas D. Rossing, Chair
Stanford Univ., Stanford, CA 94022

Chair's Introduction—7:55

Invited Paper

8:00

3aID1. Objective evaluation of musical instrument quality: A grand challenge in musical acoustics. D. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, d.m.campbell@ed.ac.uk)

Over the last few decades, increasingly sophisticated experimental and computational studies have clarified the processes involved in sound production in musical instruments. One of the principal goals of this research effort has, however, remained tantalizingly elusive: the establishment of clear and unambiguous relationships between objectively measured properties of an instrument and judgments of its musical qualities by an experienced player. This is partly because player evaluation is a subtle and highly subjective process in which many different aspects of the instrument's performance may be tested. Early studies concentrated on the steady state spectra of sound recorded in the far field of the instrument. More recently, it has been recognized that transient aspects of an instrument's performance are important in judgments of quality made by performers. These aspects include the ease with which a stable regime of oscillation can be initiated, and the flexibility with which pitch, amplitude, and timbre can be modified during performance. Attempts to define "playability" of an instrument in scientific terms, and to relate these scientific metrics to the vocabulary used by performers in judgments of playability, have been partially successful, but many questions remain unanswered.

WEDNESDAY MORNING, 5 JUNE 2013 513ABC, 8:55 A.M. TO 12:00 NOON

Session 3aAAa

Architectural Acoustics and Musical Acoustics: Virtual Concert Hall Acoustics I

Sungyoung Kim, Cochair
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Wieslaw Woszczyk, Cochair
Music Res., McGill Univ., Schulich School of Music, 527 Sherbrooke St. West, Montreal, QC H3A1E3, Canada

Chair's Introduction—8:55

Invited Papers

9:00

3aAAa1. Towards the state of the art in virtual acoustics technology. Wieslaw Woszczyk, Doyuen Ko, and Jonathan Hong (Music Res., McGill Univ., Schulich School of Music, 527 Sherbrooke St. West, Montreal, QC H3A1E3, Canada, wieslaw@music.mcgill.ca)

Active acoustics enhancement systems have been applied in room acoustics for decades yet it would be hard to say that their ultimate performance has been achieved, even with the power of today's fast digital processors. There are many challenges ahead including the creation of homogeneous directionless diffuse field using a limited number of discrete loudspeaker sources. There is also a need to create soundfield having some localized directional properties mimicking the phenomena found in acoustics of musical instruments interfacing the acoustics of rooms. Some solutions to these challenges will be presented and discussed by the authors.

9:20

3aAAa2. Active field control using sound field generation technique—Case study of a Live concert at a virtual Renaissance church. Takayuki Watanabe, Masahiro Ikeda (Spatial Audio System, Yamaha Corp., 10-1 Nakazawa-cho, Nakaku, Hamamatsu 430-8650, Japan, takayuki_watanabe@gmx.yamaha.com), and Sungyoung Kim (ECTET, Rochester Inst. of Technol., Rochester, NY)

In the Renaissance area, musical culture was centered at and spread across churches. In order to appreciate Renaissance music, therefore, it is important to account for the influence of acoustics of churches at that era so that audiences today can experience homogeneous musical appreciation. We used an Active Field Control system to create the acoustics using measured the impulse responses (IRs) of a church. The system consists of directional microphones, head amps, a convolution engine, a matrix processor, amplifiers, and loudspeakers. And it picks up the direct response of performance, convolves it with the measured IRs, and reproduces the resulting sound using loudspeakers around the room. Loudspeaker positions in the performance hall are equivalent to the positions where the IRs had been measured at the church. This technique allows us to convincingly recreate not only the reverberation time but also the spatial impressions of the church at the performance hall. In addition, we modified the IRs so that the inherent acoustical characters of the performance space would less influence to recreation of the target church acoustics. We used a 6-channel recording/reproduction system to evaluate the modification to the IRs and the impressions of the recreated acoustics in the room. This paper presents the results of the experiment.

9:40

3aAAa3. Optimizing acoustics for spoken word using active acoustics. Steve Ellison and Pierre Germain (Meyer Sound Labs., Inc., 2832 San Pablo Ave., Berkeley, CA 94702, ellison@meyersound.com)

Teleconferencing, classrooms, lectures, drama, and worship all rely on spoken word to convey a message. The successful receipt of the message is largely dependent on the acoustic of the room, the vessel for the message, both in amplified and unamplified situations. A room that supports teleconferencing well will have minimal early reflections and reverberation, yet the same room may be used in a classroom environment that would benefit from early reflections. Active acoustic systems can be used to provide this acoustic energy. Early Reflection Benefit (ERB) will be revisited, and active acoustic systems utilized for speech in various contexts will be described.

10:00

3aAAa4. Evaluation of stage acoustics preference for a singer using oral-binaural room impulse responses. Luis A. Miranda Jofre, Densil A. Cabrera, Manuj Yadav (Faculty of Architecture, Design and Planning, Univ. of Sydney, 5/27 Fisher St., Petersham, Sydney, NSW 2049, Australia, lmir9852@uni.sydney.edu.au), Anna Sygulska (Faculty of Architecture, Poznan Univ. of Technol., Poznan, Greater Poland, Poland), and William L. Martens (Faculty of Architecture, Design and Planning, Univ. of Sydney, Sydney, NSW, Australia)

There are two main objective measurement methods in current practice that can be used to evaluate the stage acoustic conditions for singers. One is the stage support metrics (ST_{Early} and ST_{Late}, included in the standard ISO 3389-1), and the other is the voice support metrics proposed by Pelegrín-García (room gain (G_{RG}) and voice support (ST_V)). All of these metrics use energy integration from impulse responses to derive the acoustic descriptors. This overlooks two potentially important features of the responses: the temporal distribution of the impulse response within the evaluation period, and the directional distribution for the spatial impulse response within the evaluation period. In this paper, a method to study the effect of these features is proposed and tested. This method is based on the auralization of ones' own voice in rooms using oral-binaural room impulse responses (OBRIRs). The OBRIRs used are created by combining synthesized early reflections with a recorded reverberant tail. The early reflections are manipulated in their arrival time, arrival direction, and strength. Results of a pilot study indicate that a wide range of on-stage acoustic quality can be achieved for responses showing a given ST_{Early} value due to variation in the temporal and spatial distribution of reflected energy.

10:20

3aAAa5. Augmented stage support in ensemble performance using virtual acoustics technology. Doyuen Ko, Wieslaw Woszczyk, and Jonathan Hong (Sound Recording, Music Res., McGill Univ., Schulich School of Music, 555 Sherbrooke west, Montreal, QC H3A 1E3, Canada, doyuen.ko@mcgill.ca)

Perceptual evaluation of electronically varied acoustic conditions has been perfomed with 15 musicians in a chamber orchestra rehearsal. Using Virtual Acoustics Technology (VAT), a room impulse response based active acoustic enhancement system, four different acoustic conditions were presented to the musicians. Condition 1 was the natural room itself without the VAT system. Condition 2 had a moderate level of VAT enhancement with approximately 10% increased EDT value from the condition 1. Condition 3 duplicated the condition 2 except for another 10% rise of EDT, and condition 4 offered enhancement utilizing multiple early reflections IRs without extending the reverb time of the space. The conditions were presented to the ensemble in random order and the loudness of all four conditions was carefully matched within 1 dB variance. Evaluation results indicated that 65% of participating musicians prefered condition 4 and their preference was highly correlated with perceptual attributes such as "feeling of stage support," "feeling of intimacy," and "quality of reverberation." The objective measurements also confirmed the improvements in stage acoustics support parameters (ST1 and ST2) in condition 4.

10:40

3aAAa6. Sound Cask: Music and voice communications system with three-dimensional sound reproduction based on boundary surface control principle. Yusuke Ikeda and Shiro Ise (Faculty of Eng., Kyoto Univ., Yasaka Shijo building 6F, Tateuri nakano-cho 106, shimogyou-ku, Kyoto-shi, Kyoto-fu 6008006, Japan, ae-yusuke-ikeda@archi.kyoto-u.ac.jp)

To reproduce a highly realistic sound field reproduction, we have developed a 3-D sound reproduction system based on the boundary surface control (BoSC) principle. We set up an Internet connection among the systems, which enables distant speakers to carry out voice telecommunication by simultaneously hearing the same sound field and sensing the other speakers' simulated positions as if they were in the same location. In this system, a listener can freely move his head, because this system reproduce the sound not only at points near his ears but also an area around his head. In this paper, we introduce the "Sound Cask," which is a 96-channel sound reproduction system

based on the BoSC principle. The Sound Cask is used to realize musical telecommunication as if the performers were all playing in the same hall. The system provides a space large enough to play a small musical instrument such as a violin, which allows listening and communication accompanied with the natural body movements of the performer. Another specific feature of the system is that loudspeakers are installed in every possible direction except the floor. The system is designed to be portable so that it can be assembled or disassembled when needed.

11:00

3aAAa7. Subjective evaluation of a virtual acoustic system: Trials with three-dimensional sound field reproduced by the "Sound Cask". Maori Kobayashi (Meiji Univ., 4F Bell Shimokitazawa, 2-36-9 Kitazawa, Setagaya, Tokyo 155-0031, Japan, te11001@meiji.ac.jp), Kanako Ueno (Meiji Univ., Kanagawa, Japan), Mai Yamashita, Shiro Ise (Kyoto Univ., Kyoto, Japan), and Seigo Enomoto (National Inst. of Information and Commun. Technol., Kyoto, Japan)

It has been necessary to establish subjective measures for the performance of the virtual acoustic systems. In this paper, we report our trials to evaluate the performance of a three-dimensional sound field reproduction system based on the boundary surface control principle, the "Sound Cask." First, we introduce our investigations for the experts of audio engineering in order to clarify the difference of auditory impression between the Sound Cask and conventional audio systems. Second, we report psychological and physiological experiments focusing on the advantageous points of the Sound Cask, localization performance, and a clear sense of reality, which were pointed out in the investigations for the experts. Finally, we discuss the issues to be considered for subjective evaluation of virtual acoustic systems for future studies.

Contributed Papers

11:20

3aAAa8. Modeling and simulation of Gamelan Bali concert hall based on objective acoustic parameters. Ni Putu Amanda Nitidara, I G. Nyoman Merthayasa, and Joko Sarwono (Eng. Phys., Bandung Inst. of Technol., Program Studi Teknik Fisika Institut Teknologi Bandung Jalan Ganesha 10 Gedung Lab. Teknologi VI Lantai 2, Bandung, West Java 40132, Indonesia, amandanitidara@yahoo.co.id)

Gamelan Bali music performances require special place to highlight the quality of the acoustic performances. A concert hall dedicated for Gamelan Bali was proposed to perform a better acoustic quality. Studies on Gamelan Bali has been done to achieved an optimum value of sound fields in Gamelan Bali concert hall. This paper shows a geometrical model designed to fulfill the suitable sound fields of Gamelan Bali Concert Hall. The model was modified from shoebox shaped with rear and side balconies. The acoustic performances of the model at 1 kHz summarized as follows: Tsub = 1.41 s, LL = 80.2 dBA, $\Delta t1 = 39.43$ ms, and IACC = 0.33. Those values meet the optimum values from the result of previous studies. Auralization of sound in the room was also done for the purpose of subjective judgment.

11:40

3aAAa9. Electronic architecture—Recent developments in the design, implementation, and performance of time variant acoustic enhancement systems. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

When an electro-acoustic enhancement system is integrated with architectural surfaces in an enclosed volume, the acoustical conditions experienced throughout the venue result from the interaction between the two. Over two decades, we have encountered performance venues with a variety of sizes, shapes, and configurations. Each venue presents unique challenges in configuring architectural elements, as well as the makeup of the electro-acoustic enhancement system, in order to meet the expectations of the users and the audience. This paper describes important elements germane to a successful outcome—from the essential qualities of system components, to the physics and physiology that enable humans to perceive sonic quality. Examples of performance spaces with unique physics and system integration will be discussed.

WEDNESDAY MORNING, 5 JUNE 2013 513DEF, 9:00 A.M. TO 11:40 A.M.

Session 3aAAb

Architectural Acoustics: Architectural Acoustics Potpourri

Roger M. Ellingson, Chair
RM Ellingson Design & Development, LLC, 8515 SW Barnes Rd., Portland, OR 97225

Contributed Papers

9:00

3aAAb1. Assessing the acoustics of offices: A case of standardization. Marc Asselineau (Acoustics, PEUTZ & Assoc., 10 B rue des Messageries, Paris F75010, France, m.asselineau@peutz.fr)

Standards are supposed to help promote better understanding between the various interested parties. As such they can help an end user understand what the acoustic performances of his future premises will be… as long as such a standard exists. Over the years, various standards have been developed by several countries to characterize the acoustics of non partitioned offices. The tactics developed by ISO follow the Scandinavian standards trend with emphasis on the spatial sound level decay in the furnished space while the north American standards are more concerned with speech discretion. Are those tactics really opposed, or are they complementary? This paper intends to submit a comparison of a few standards (ISO, NF, CAN) applied on a couple of representative cases. It turns out that such standards represent the habits and practice of those who wrote it.

9:20

3aAAb2. Subjective experiment on suitable speech-rate of public address announcement in public spaces. Sakae Yokoyama (I.I.S., The Univ. of Tokyo, Komaba 4-6-1, Meguro-ku 153-8505, Japan, sakae@iis.u-tokyo.ac.jp) and Hideki Tachibana (Chiba Inst. of Technol., Narashino, Japan)

In such public spaces as railway stations, airport terminal buildings, shopping arcades, etc., it is often the case that information provided through public address (P.A.) system is deteriorated by the influences of reverberation and background noise. This problem is serious especially for the case of providing various information and emergency evacuation alarms in the case of such disasters as earthquake and fire. Regarding this acoustic problem in public spaces, the authors performed field surveys in various public spaces, in which actual P.A. announcements were recorded and reproduced in an anechoic room by applying the 6-channel recording/reproduction technique to realize 3D aural impression. As a result of the subjective experiments on speech intelligibility (ease in listening) performed in the simulated sound field, it has been found that the speed of speech is an essential condition as well as reverberation and noise level. To investigate the way of improving speech intelligibility in such spaces, subjective experiment was performed by changing the reverberation time, background noise level, and the speech rate in Japanese announcements at several steps. In the experiment, a Text-To-Speech (TTS) software was used to change the speech rate. As the test subjects, students from abroad participated in the experiment and the difference of the effects of the adverse conditions between natives and non-natives was also investigated.

9:40

3aAAb3. Design of the new public address system for the cathedral of Münster, Germany. Gottfried Behler and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustraße 50, Aachen D-52056, Germany, gkb@akustik.rwth-aachen.de)

One of the most renowned cathedrals in Germany, the Dom St. Paul in Münster was completely closed for renovation for almost one year. During this time, the entire electro acoustical sound reinforcement system has been renewed. As for most buildings of this type, the acoustical situation is far away from optimal. This mainly is due to a huge reverberation time, which makes the understanding of spoken words almost impossible. Moreover, the old concept of sound reinforcement by using distributed loudspeaker systems all over the church is not satisfying anymore with respect to nowadays demands for quality and speech intelligibility. Due to the situation that the number of people in the cathedral during service times is varying from only some hundreds to over 2500 a more exible PA system is required, that takes into account that only occupied areas inside of the cathedral should be supplied with amplified sound. To achieve the target, an entirely new concept for the sound reinforcement based on digital signal distribution and modern digitally operated array loudspeakers was developed. The requirement for the speech intelligibility was to reach at least an STI of more than 0.5. The system will be discussed and results will be shown.

10:00

3aAAb4. Acoustical design of Turkish Religious Affairs Mosque. Zühre Sü Gül (R&D, MEZZO Studyo LTD, METU Technopolis KOSGEB-TEKMER No112, ODTU Cankaya, Ankara 06800, Turkey, zuhre@mezzostudyo.com) and Mehmet Čalışkan (Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

The new Turkish Religious Affairs (DIB) Mosque with its prayer capacity and outstanding volume is the largest classical-contemporary mosque project of the past decade built in Ankara, Turkey. The Mosque is also one of unique examples of its scale for which the room acoustic design is applied in its design phase. Acoustical design of DIB Mosque is critical considering speech and music related activity patterns held in such religious spaces. Interior surface forms and materials of walls and pendentives, floor finishing alternatives, geometry, and finishing of the dome are studied simultaneously with the architectural design as early as in the concept phase. Impedance tube is used for testing alternative materials for specifying sound absorption characteristics of reliefs and perforations. Computer simulation is applied as an acoustical design tool and estimations are held by ODEON v.11.23. Objective acoustical parameters including reverberation time, speech transmission index, and A-weighted sound levels are assessed with and without sound reinforcement systems for fully and partially occupied mosque conditions. Auralizations are held for imam and muezzin in different forms of religious call out to prayers. Evaluation of the space indicates that the optimized acoustical field is proper for intended functions of use in a mosque and satisfies desired tranquil environment.

10:20–10:40 Break

10:40

3aAAb5. Design and experience with subterranean installation of a fully anechoic acoustical testing chamber. Roger M. Ellingson (RM Ellingson Design & Development, LLC, 8515 SW Barnes Rd., Portland, OR 97225, Rogere@Rmegen.net) and Patrick V. Helt (National Ctr. for Rehabilitative Auditory Res., Portland, OR)

One of the purposes of a fully anechoic chamber is to provide a very quiet, near-echo-free environment simulating free-field acoustical conditions. From design specification to completion, installation of a fully anechoic chamber can be an enormous undertaking as compared to installation of conventional sound-attenuated acoustical test rooms, which generally are smaller in physical size and have less stringent sound attenuation requirements. The authors had the opportunity to specify the requirements and oversee the installation of a fully anechoic chamber designed to support near-full-frequency, human-hearing range acoustical experiments at the VA National Center for Rehabilitative Auditory Research (NCRAR), located in Portland, OR, USA. Design and installation of NCRAR's chamber to support entry at laboratory floor level was complicated by the job site location, a subterranean area beneath a structure that accommodates clinical and research offices on the upper floors, and serves as a parking garage on the lower floors. In addition to ambient noise considerations, existing architectural restrictions included parking traffic patterns, below-grade earthquake beams, limited overhead clearance, ground water seepage, and storm water flow patterns. The purpose of this paper is to share pictorial-illustrated experiential results on specification, design, installation, and chamber operation as well as architectural considerations, site preparation, and construction detail.

11:00

3aAAb6. A six sensor method for measuring acoustic properties in ducts. Timothy J. Newman, Anurag Agarwal, Ann P. Dowling (Dept. of Eng., Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, United Kingdom, tjn25@cam.ac.uk), Ludovic Desvard, and Ryan Stimpson (Aeroacoustic Res. Team, Dyson Ltd., Malmesbury, Wiltshire, United Kingdom)

An accurate description of sound propagation in a duct is important to obtain the sound power radiating from a source in both near and far fields. A technique has been developed and applied to decompose higher-order modes of sound emitted into a duct. Traditional experiments and theory based on two-sensor methods are limited to the plane-wave contribution to the sound field at low frequency. Due to the increase in independent measurements required, a computational method has been developed to simulate sensitivities of real measurements (e.g., noise) and optimize the set-up. An experimental rig has been constructed to decompose the first two modes using six independent measurements from surface, flush-mounted microphones. Experiments were initially performed using a loudspeaker as the source for validation. Subsequently, the sound emitted by a mixed-flow fan has been investigated and compared to measurements made in accordance with the internationally standardized in-duct fan measurement method. This method utilizes large anechoic terminations and a procedure involving averaging over measurements in space and time to account for the contribution from higher-order modes. The new method does not require either of these added complications and gives detail about the underlying modal content of the emitted sound.

11:20

3aAAb7. Acoustics evolution of the Moscow Conservatory Great Hall after renovation in 2011. Nikolay Kanev and Anatoly Livshits (Acoust. Group, 4, Svernika st., Moscow 117036, Russian Federation, nikolay.kanev@mail.ru)

The Great Hall of the Moscow P.I. Tchaikovsky Conservatory is one of the best concert halls in Russia. Its acoustics is appreciated very much by both musicians and audience. In 2011, the Great Hall was renovated, after renovation its acoustics conditions remained at very high level that was confirmed by means of objective impulse response measurements and subjective estimations. In order to control how acoustics conditions are changing with time the observation of main acoustics parameters is carrying out with half year gaps. In this work we present the results of four measurements fulfilled in June 2011 (just after the renovation), December 2011, June 2012, and December 2012. The reverberation time decreases at low frequencies, whereas it is stable at middle frequencies. Periodic variations of the reverberation time take place at high frequencies. These variations are probably connected with seasonal changes of temperature and humidity conditions in the hall. Changes of other acoustics parameters correlate to reverberation time.

WEDNESDAY MORNING, 5 JUNE 2013 510B, 8:55 A.M. TO 12:00 NOON

Session 3aAB

Animal Bioacoustics and Psychological and Physiological Acoustics: Perceiving Objects I

Caroline M. DeLong, Cochair
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Eduardo Mercado, Cochair
Dept. of Psych., Univ. at Buffalo, Buffalo, NY 14260

Chair's Introduction—8:55

Invited Papers

9:00

3aAB1. Human performance in aural classification of sonar echoes. Nancy Allen and Paul C. Hines (Defence R&D Canada - Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, nancy.allen@drdc-rddc.gc.ca)

Active sonar echoes from man-made objects and those from naturally occurring features in certain coastal environments can be difficult to distinguish when using traditional sonar processing techniques and visual displays. An approach being investigated at Defence R&D Canada-Atlantic for addressing this challenge is to exploit the capability of human hearing for discriminating between these two classes of echoes. Part of the work consisted in producing a human-performance baseline. Two human listening tests were designed and carried out. Both used stimuli that were generated from a sample of echoes recorded during actual sonar experiments at sea. In the second test, a 500-Hz high-pass filter was applied to the stimuli. Quantitative data from the rating-exercise portion of the tests were used to produce receiver-operating-characteristic (ROC) curves for modeling how well the participants could distinguish the two classes of echoes. For both tests, the results show that the listeners could hear differences, but performance was significantly better in the first test. Qualitative data collected from the questionnaire portion of the tests helped to interpret some of the performance results.

9:20

3aAB2. Recognizing objects from multiple orientations using dolphin echoes. Caroline M. DeLong, Amanda Heberle, Kayla Mata (Psychology, Rochester Inst. of Tech., 18 Lomb Memorial Dr., Rochester, NY 14623, cmdgsh@rit.edu), Heidi E. Harley (Psychology, New College of Florida, Sarasota, FL), and Whitlow W. Au (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, Kaneohe, HI)

Object constancy, the ability to recognize objects despite changes in orientation, has not been well studied in the auditory modality. Dolphins use echolocation for object recognition, and objects ensonified by dolphins produce echoes that can vary significantly as a function of orientation. In four experiments, human listeners had to classify echoes from objects ensonified with dolphin signals. Participants were trained to discriminate among the objects using an 18-echo stimulus from a 10 degree range of aspect angles, then tested with novel aspect angles across a 60 degree range. In the first two experiments, the three objects varied in material, size, and shape. Participants were typically successful recognizing the objects at all angles ($M = 78\%$). In experiment 3, the three objects had the same material but different shapes. Participants were often unsuccessful recognizing the objects at all angles ($M = 46\%$). In experiment 4, participants had to classify echoes from four fish species across a wider range of angles (330 degrees). Preliminary results show overall poor performance ($M = 45\%$). These results suggest that object characteristics play a role in whether performance is more view-dependent or view-invariant. These studies can provide insight into the process dolphins use to identify objects.

9:40

3aAB3. Exploring the capacity of neural networks to recognize objects from dolphin echoes across multiple orientations. Matthew G. Wisniewski (Psychology, Univ. at Buffalo, The State Univ. of New York, 260 Callodine Ave., Amherst, NY 14226, mgw@buffalo.edu), Caroline M. DeLong, Amanda L. Heberle (Psychology, Rochester Inst. of Technol., Rochester, NY), and Eduardo Mercado (Psychology, Univ. at Buffalo, The State Univ. of New York, Buffalo, NY)

Dolphins naturally recognize objects from multiple angles using echolocation. With training, humans can also learn to accurately classify objects based on their echoic features. In this study, we used neural networks to identify acoustic cues that enable objects to be recognized from varying aspects. In simulation 1, a self-organizing map was able to differentiate a subset of objects using only amplitude and frequency cues, but it classified some echoes from different objects as being from the same object. In simulation 2, a multilayer

perceptron was trained through error correction to identify objects based on echoes from a single aspect, and then tested on its ability to recognize those objects using echoes from different orientations. Overall, perceptrons performed similarly to trained undergraduates. Analysis of network connection weights revealed that both the amplitude and frequency of echoes, as well as the temporal dynamics of these features over the course of an echo train, enabled perceptrons to accurately identify objects when presented with novel orientations. These findings suggest that learning may strongly impact an organism's ability to echoically recognize an object from any viewpoint.

10:00

3aAB4. Object selection by head aim and acoustic gaze in the big brown bat. Jason Gaudette (NUWC, Providence, RI), Laura Kloepper, and James Simmons (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02906, laura_kloepper@brown.edu)

Echolocating bats use their active sonar to locate, discriminate, and capture flying prey. A special challenge is tracking and pursuing a discrete moving object, sometimes in cluttered surroundings. Bats rely on head aim to follow their prey's location throughout an entire capture sequence. By directing the sound emission organ (mouth or noseleaf) and thus acoustic gaze toward the target, the prey is kept on the main broadcast axis with the flattest incident spectrum. Although bats can perform head-aim tracking with an accuracy of a few degrees, we have explored the dynamics of the transmitted beam during tracking and its impact on angular precision. We measured head aim and acoustic gaze in the big brown bat (*Eptesicus fuscus*) with a 224-element microphone array. This array allows for fine scale, independent measurements of the beam across many frequencies with a high signal-to-noise ratio. Bats were trained to track moving real and electronic targets, and the head aim and acoustic gaze were recorded. We specifically examined the possibility that bats move different frequencies of their beam at different angular rates, with particular attention to the harmonics of the broadcasts. [Work supported by ONR, NSF, and NUWCDIVNPT.]

10:20

3aAB5. Temporal signal processing of dolphin biosonar echoes from fish prey. Whitlow Au and Hui Ou (Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, wau@hawaii.edu)

Dolphins that emit whistle signals (except for sperm whales) project short broad- band biosonar clicks containing about 5 to 7 cycles with exponential decaying envelope and Q (center frequency over bandwidth) between 2 and 3. The broadband nature of the biosonar clicks allow for good temporal resolution of echo highlights, which in turn allows for the discriminations of different targets including fish prey. Most of the echoes from fish originate from signals reflecting off the swim bladder of fish. Different species of fish have swim bladders of different shape, size, and orientation so that echoes from these species can often be discriminated from temporal cues. The echoes contain many highlights as the signals reflect off different surfaces and parts of the fish body and swim bladder. This presentation will discuss the temporal characteristics of echoes from fish prey, which are highly aspect dependent and will discuss the six temporal parameters that were used in a support vector machine (SVM) to discriminate between species. Results suggest how dolphins can classify fish based on their echoes and provide some insight as to which features might enable the classification.

10:40

3aAB6. Dolphin strategies for long-range object detection and change detection. James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil)

Dolphin strategies for detecting objects and changes in objects were investigated by having three trained bottlenose dolphins perform long-range echolocation tasks. The tasks featured the use of "phantom" echoes produced by capturing the dolphin's outgoing echolocation clicks, convolving the clicks with the impulse response of a physical target to create an echo waveform, then broadcasting the delayed, scaled echo waveform back to the dolphin. Dolphins were trained to report the presence of phantom echoes or a change in phantom echoes. Target simulated ranges varied from 25 to 800 m. At ranges below 75 m, all dolphins followed a single click-echo paradigm, where inter-click intervals exceeded the two-transit time (i.e., the dolphins waited to receive the echo from a click before emitting the next click). As the range increased beyond 75 m, two of the three dolphins increasingly produced bursts, or "packets," of several clicks, then waited for the packet of echoes to return before emitting another packet of clicks. The third dolphin instead utilized very high click repetition rates. The use of click packets may be a response to a limitation in the dolphin's ability to employ multi-echo processing with large inter-echo delays.

11:00

3aAB7. Dolphin echolocation is not seeing with sound. Heidi E. Harley (Psychology, New College of Florida, Div. of Social Sci., 5800 Bay Shore Rd., Sarasota, FL 34243, harley@ncf.edu), Wendi Fellner, and Barbara Losch (The Seas, Epcot®, Walt Disney World® Resorts, Kissimmee, FL)

Dolphin echolocation is often described as "seeing with sound;" however, vision and audition vary substantially in terms of direct access to spatial information. This study investigated spatial representation based on echoic information by having a dolphin match objects that varied only in shape. Stimulus sets (3 objects each) included unfamiliar objects made from (1) the same PVC parts (controlled sets), (2) different PVC parts (uncontrolled sets), or (3) non-PVC junk objects (junk sets). Sets were presented for 5 18-trial sessions in each of two conditions: an echoic condition (objects underwater with dolphin eyecups occluding vision) and a visual condition (objects in air). Performance accuracy varied across set type and condition. Worst was echoic controlled (12 sets, M = 45%) followed by echoic uncontrolled (12 sets, M = 50%). Visual performance was significantly better: controlled (M = 67%), uncontrolled (M = 76%). Echoic performance ranged from 32% to 86% across sets; visual performance from 50% to 100%. In contrast, performance accuracy with junk sets was higher echoically (2 sets, M = 81%) than visually (2 sets, M = 58%). Shape does not appear to be easily accessible to dolphins via echolocation, although it is accessible through vision. Dolphins integrate information about objects across modalities; they likely gain most shape information through vision.

11:20

3aAB8. Auditory object formation in Cope's gray treefrogs (*Hyla chrysoscelis*). Mark A. Bee (Ecology, Evolution and Behavior, Univ. of Minnesota, 100 Ecology, 1987 Upper Buford Circle, St. Paul, MN 55108, mbee@umn.edu) and Katrina M. Schrode (Grad. Program in Neurosci., Univ. of Minnesota, St. Paul, MN)

Hearing and acoustic communication in "real world," multi-source environments require animals to group sound elements produced by the same source into perceptually coherent "auditory objects." However, research on nonhuman animal communication rarely investigates perceptual processes involved in forming auditory objects of communication sounds. We tested the hypotheses that spectral and spatial proximity promote the sequential integration of temporally separated sounds produced by the same source into coherent auditory objects of acoustic signals. Male gray treefrogs produce a pulsatile advertisement call; females prefer longer calls (= more pulses) to shorter calls and discriminate against calls missing pulses. We gave females a choice between a short but spectrally and spatially coherent call (25 pulses) and a longer call (35 pulses) in which alternating groups of 5 pulses had different frequencies (ΔF, 0–12 semitones) and came from different locations ($\Delta\theta$, 0° or 90°). Females generally preferred the longer call at smaller values of ΔF and $\Delta\theta$, indicating a role for spectral and spatial proximity in sequential integration. Under some conditions, however, subjects showed a surprising willingness to integrate pulses despite large ΔFs. Together, these data shed light on the perceptual cues that receivers exploit to form coherent auditory objects of communication sounds.

11:40

3aAB9. Auditory scene analysis in budgerigars (*Melopsittacus undulatus*) and zebra finches (*Taeniopygia guttata*). Micheal L. Dent, Erikson G. Neilans, Mary M. Flaherty, and Amanda K. Martin (Psychology, Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260, mdent@buffalo.edu)

Deciphering the auditory scene is a problem faced by humans and animals alike. However, when faced with overlapping sounds from multiple locations, listeners are still able to attribute the individual sound objects to their individual sound-producing sources. Here, we determined which characteristics of sounds are important for streaming versus segregating in birds. Budgerigars and zebra finches were trained using operant conditioning procedures on an identification task to peck one key when they heard a whole zebra finch song and to peck another when they heard a zebra finch song missing a middle syllable. Once the birds were trained to a criterion performance level on those endpoint stimuli, probe trials were introduced on a small proportion of all trials. The probe songs contained modifications of the incomplete training song's missing syllable. When the bird responded as if the probe was a whole song, it suggests they streamed together the altered syllable and the rest of the song. When the bird responded non-whole song, it suggests they segregated the altered probe from the rest of the song. Results show that some features, such as spectrotemporal similarity and location, are more important for streaming than other features, such as timing.

WEDNESDAY MORNING, 5 JUNE 2013 510D, 10:55 A.M. TO 11:40 A.M.

Session 3aAO

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

John A. Colosi, Chair
Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943

Chair's Introduction—10:55

Invited Paper

11:00

3aAO1. Acoustical tomography in the shallow water ocean: Dream or reality? Philippe Roux (ISTerre, CNRS - Université Grenoble 1, Maison des Géosciences, Rue de la piscine, Grenoble 38041, France, philippe.roux@obs.ujf-grenoble.fr)

Acoustical tomography in shallow waters relies on the identification and tracking of stable ray arrivals. The variation of the arrival time of these rays is used to solve the inverse problem and to estimate the physical properties as sound speed variations associated to internal waves or currents. In practice, however, technical difficulties appear (1) when the number of resolved eigenrays in this multipath environment is too small using a set of point-to-point recordings or (2) when the travel-time fluctuations are dominated by the fast-evolving surface that blur the slower internal-wave driven perturbations around the thermoclyne. Recent experimental studies based on laboratory-scaled demonstrators showed the efficiency of array processing algorithms in combination with source-receiver arrays to tackle issue (1) above. Depending on the waveguide geometry, the acquisition procedure can also be adjusted to provide a separation between the slow and fast travel-time fluctuations. Inversion results based on laboratory experiments are presented that focus on the dynamics of (1) a gravity wave at the air-water interface and (2) a thermal plume in the water column. Finally, these results are discussed in the framework of ocean acoustic research.

Session 3aBAa

Biomedical Acoustics: Delivery of Nucleic Acids (DNA, siRNA, antisense oligos)

Tyrone M. Porter, Cochair
Boston Univ., 110 Cummington St., Boston, MA 02215

Raffi Karshafian, Cochair
Dept. of Phys., Ryerson Univ., 350 Victoria St., KHE 329E, Toronto, ON M5B 2K3, Canada

Invited Papers

9:20

3aBAa1. Ultrasound-mediated gene delivery – Cardiovascular applications for Chronic ischemia, heart failure, and ischemia-reperfusion injury. Howard Leong-Poi (Cardiology/Medicine, St. Michael's Hospital, 6-044 Queen Wing, 30 Bond St., Toronto, ON M5B1W8, Canada, leong-poih@smh.ca)

Ultrasound-mediated gene delivery (UMGD) is a non-invasive gene transfer technique, utilizing high power ultrasound and DNA-bearing microbubbles. Despite modest transfection efficiency, its high organ, tissue specificity, and repeatability make it an attractive therapeutic option. UMGD has been used in a variety of *in vivo* applications, including cardiac and skeletal muscle, kidney, liver, cerebral, and even lung, and have been studied using many gene vectors, including plasmid, viral, and small interfering RNA. This presentation will focus specifically on cardiac applications using plasmid DNA, including (1) introduction of UMGD in the heart, including optimization of parameters and protocols, (2) UMGD for therapeutic angiogenesis in chronic ischemia, including multi pro-angiogenic gene therapy and combination gene- and progenitor cell-based therapies for chronic hindlimb ischemia, and (3) applications for anti-apoptotic therapy in heart failure and ischemia-reperfusion injury.

9:40

3aBAa2. Effective ultrasound-targeted microbubble destruction-mediated gene transfer into the livers of small and large animals. Carol H. Miao, Misty L. Noble, Shuxian Song, Ryan R. Sun (Pediatrics, Seattle Children's Res. Inst. and Univ. of Washington, 1900 Ninth Ave. C9S-7, Seattle, WA 98125, miao@uw.edu), Christian S. Kuhr (Benaroya Res. Inst., Seattle, WA), Scott S. Graves (Fred Hutchinson Cancer Res. Ctr., Seattle, WA), George W. Keilman, Kyle P. Morrison (Sonic Concepts Inc., Seattle, WA), Keith R. Loeb (Fred Hutchinson Cancer Res. Ctr., Seattle, WA), Andrew A. Brayman, Marla Paun (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Rainer F. Storb (Fred Hutchinson Cancer Res. Ctr., Seattle, WA), and Samuel S. Sun (Pediatrics, Seattle Children's Res. Inst. and Univ. of Washington, Seattle, WA)

Ultrasound (US)-targeted microbubble (MB) destruction (UTMD) can significantly enhance gene delivery in mouse livers when pDNA/MBs were injected into the portal vein (PV) with simultaneous US exposure using a focused transducer. However, this transducer was ineffective in enhancing gene transfer into rats. A 13-mm diameter unfocused transducer was designed and the delivery route of pDNA/MBs was modified into a specific liver lobe, resulting in $>$100-fold increase in luciferase expression in rats. To facilitate the translation into human application, many technical issues were explored in large animal models. We applied 1.1 MHz US to a targeted canine liver lobe with simultaneous injection of pDNA/MBs into a PV segmental branch and occlusion of the inferior vena cava. For more effective treatment of large tissue volumes, a 52-mm planar unfocused transducer was specifically constructed. Its apodized dual element configuration greatly reduced the near field transaxial pressure variations, producing a uniform field of US exposure for the treated tissues. Together with a 15 kW-capacity US amplifier, a 692-fold increase of gene expression in canines was achieved at 2.7-MPa. Transaminase levels and histology analysis indicated minimal tissue damage. These results demonstrated that UTMD is highly promising for safe and efficient gene delivery into the liver.

10:00

3aBAa3. Non-invasive delivery of small interfering ribonucleic acid for reduction of *Huntingtin* expression in the brain is achieved using focused ultrasound to disrupt the blood-brain barrier. Alison Burgess, Yuexi Huang (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada), William Querbes, Dinah W. Sah (Alnylam Pharmaceuticals Inc., Cambridge, MA), and Kullervo Hynynen (Med. Biophys., Univ. of Toronto, Sunnybrook Health Sci. Ctr., Toronto, ON M4N 3M5, Canada, khynynen@sri.utoronto.ca)

Huntington's disease is caused by a mutation in the *Huntingtin* (Htt) gene, which leads to neuronal dysfunction and cell death. Silencing of the Htt gene can halt or reverse the progression of the disease indicating that RNA interference is the most effective strategy for disease treatment. However, small interfering RNA (siRNA) does not cross the blood-brain barrier and therefore delivery to the brain is limited. Here, we demonstrate that focused ultrasound (FUS), combined with intravascular delivery of microbubble contrast agent, was used to locally and transiently disrupt the BBB in the right striatum of adult rats. 48 hours following treatment with siRNA, the right (treated) and left (control) striatum was dissected and analyzed for Htt mRNA levels. We demonstrate that FUS can non-invasively deliver siRNA-Htt directly to the striatum leading to a significant reduction of Htt expression in a dose dependent manner. Furthermore, we show that reduction of Htt with siRNA-Htt was greater when the extent of BBB disruption was increased. This study demonstrates

that siRNA treatment for knockdown of mutant Htt is feasible without the surgical intervention previously required for direct delivery to the brain. Non-invasive delivery of siRNA through the blood-brain barrier (BBB) would be a significant advantage for translating this therapy to HD patients.

10:20

3aBAa4. Polyplex-microbubbles for improved ultrasound-mediated gene therapy. Mark Borden, Shashank Sirsi (Mech. Eng., Univ. of Colorado, 1111 Eng. Dr., Campus Box 427, Boulder, CO 80309, mark.borden@colorado.edu), Sonia Hernandez, Shunichi Homma (Oncology, Columbia Univ., New York, NY), Jessica Kandel (Surgery, Columbia Univ., New York, NY), and Darrell Yamashiro (Oncology, Columbia Univ., New York, NY)

Sonoporation is an established method whereby acoustically stimulated microbubbles create pores in the endothelium to promote the delivery of nucleic acids and other macromolecules through the vasculature. In this presentation, we describe the development of polyplex-microbubbles to enhance delivery and intracellular trafficking to target cells beyond the endothelium following sonoporation. Our design combines the tissue targeting capability of microbubbles with the cellular targeting capability of polyplexes, which are self-assembled particles comprising nucleic acids and a cationic polymer. The first purpose of the cationic polymer (polyethylenimine, PEI) is to condense and protect the nucleic acids on the microbubble surface as it travels in circulation from the site of injection to the target tissue. Polyethylene glycol (PEG) is conjugated to the PEI to reduce immunogenicity. Ultrasound is applied to the target tissue to fragment the microbubbles and release the polyplexes and to porate the endothelium, allowing the polyplexes to extravasate. The second purpose of PEI is to promote endocytotic uptake and subsequent endosomal escape of the nucleic acids into the cytoplasm by the so-called "proton-sponge" effect. Here, we will describe polyplex-microbubble fabrication and characterization, as well as *in vivo* testing. Our results indicate that this is a promising design for cancer gene therapy.

10:40

3aBAa5. Small interfering ribonucleic acid delivery with phase-shift nanoemulsions. Mark T. Burgess and Tyrone M. Porter (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, marktb@bu.edu)

Acoustic cavitation offers a unique approach to small interfering RNA (siRNA) delivery compared to current methods. Typically, preformed microbubbles are used as cavitation nuclei to permeabilize cells and facilitate siRNA entry into the cytoplasm. However, microbubbles are restricted to the vasculature space and suffer from stability issues that limit their applicability. Alternatively, phase-shift nanoemulsions (PSNE) possess the long circulation and extravasation properties of nanoparticles, while also serving as cavitation nuclei in tissue upon acoustic droplet vaporization. Here we report the use of PSNE for delivery of siRNA engineered to knockdown green fluorescent protein (GFP) expression. A cell suspension (5×10^6 cells/mL) of GFP expressing breast adenocarcinoma cells was exposed to 5 MHz pulsed ultrasound (4 MPa peak negative pressure, 3 cycles, 250 Hz, 100 s exposure duration) in the presence of PSNE ($\sim 10^9$/mL) and free siRNA (1.8 μM). Flow cytometry was used to quantify GFP expression and cell viability. There was 20% ($p < 0.05$, $n = 6$) reduction in GFP fluorescence for cells treated with GFP siRNA and 80% (+/− 6.4%) cell survival. This work highlights the potential for PSNE to serve as interstitial cavitation nuclei for siRNA delivery in tissue. [Work supported in part by NIH grants R25CA153955 and R03EB015089.]

3a WED. AM

11:00

3aBAa6. Physical mechanisms responsible for bubble translation near an interface. Daniel R. Tengelsen, Todd Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, danieltengelsen@gmail.com)

Previous models and experiments have shown that direction of bubble translation near a viscoelastic layer depends on both the standoff distance of the bubble and the elastic properties of the layer. Here the individual forces due to the incident sound field and the field reflected from the viscoelastic layer are shown to compete with one another and ultimately determine the direction of bubble translation. In addition, many other factors pertinent to the direction of bubble translation such as the incident acoustic waveform, the phase and propagation direction of the incident field, and the radial bubble dynamics are considered. The force due to the viscoelastic layer is calculated using a Green's function, which takes into account elastic waves and viscosity in the layer and the viscous boundary layer at the solid-liquid interface. [Work supported by the ARL:UT McKinney Fellowship in Acoustics and NIH DK070618.]

Contributed Paper

11:20

3aBAa7. Ultrasonic bubbles in microfluidics for red blood cells, bacteria, and yeast lysis. Siew-Wan Ohl, Tandiono Tandiono (Inst. of High Performance Computing, 1 Fusionopolis Way, #16-16 Connexis North, Singapore 138632, Singapore, ohlsw@ihpc.a-star.edu.sg), Dave Siak-Wei Ow (Bioprocessing Technol. Inst., Singapore, Singapore), Evert Klaseboer (Inst. of High Performance Computing, Singapore, Singapore), Andre Boon-Hwa Choo (Bioprocessing Technol. Inst., Singapore, Singapore), and Claus-Dieter Ohl (School of Physical and Mathematical Sci., Nanyang Technolog. Univ., Singapore, Singapore)

A custom microfluidic ultrasound device has been designed and implemented to efficiently create oscillating ultrasonic bubbles. These bubbles interact with the red blood cells, bacteria (*Escherichia coli*), and yeast (*Pichia pastoris*). Observations using high speed photography show that the red blood cells were strongly stretched. A numerical model using the Boundary Element Method is used to simulate the oscillating bubble's interaction with a nearby elastic pocket (a red blood cell model). Complex dynamics are discussed. Our ultrasonic microfluidic device is efficient in lysing *E. coli* and yeast cells for the harvesting of active intracellular content. Complete lysis of *E. coli* takes only 0.4 s while it takes about 1 s for the yeast cell. Temperature rise is minimal. We perform fluorescent intensity measurement and qRT-PCR (real-time polymerase chain reaction) to show that the functional integrity of the proteins and DNA is preserved.

Session 3aBAb

Biomedical Acoustics: Generation and Detection of High Intensity Focused Ultrasound Lesions

Ashwinkumar Sampathkumar, Chair
Biomedical Eng., Riverside Res. Inst., 156 William St., #9, New York, NY 10038

Contributed Papers

9:20

3aBAb1. Variations of temperature distribution and lesion formation induced by tissue inhomogeneity for therapeutic ultrasound. Dong Zhang (Inst. of Acoust., Hankou 22, Nanjing 210093, China, dzhang@nju.edu.cn)

High intensity focused ultrasound (HIFU) has shown potential applications in therapeutic medical fields. This work reported both theoretical and experimental investigations on the influence of tissue inhomogeneity on the temperature distribution and tissue lesion formation using the phase screen model. The inhomogeneous tissue is considered as a combination of a homogeneous medium and a phase aberration screen. Numerical simulations were performed by using the nonlinear acoustic equation and bio-heat transfer equation. Four polyethylene (PE) plates with various standard deviations were made to mimic the inhomogeneity induced by human body abdominal. Temperature rise and lesion formation induced by HIFU were measured and compared with the numerical calculations. Results indicate that the standard deviation is associated with the focusing capability and temperature field observably. This study provides a theoretical and experimental basis for the development of precise HIFU treatment plan.

9:40

3aBAb2. Thermal ablation by high-intensity-focused ultrasound using a toroidal transducer for the treatment of colorectal liver metastases during an open procedure. Clinical results. David Melodelima, Jeremy Vincenot (LabTAU - U1032, INSERM, 151 cours Albert Thomas, Lyon 69003, France, David.Melodelima@inserm.fr), Yao Chen, Aurelien Dupre, Michel Rivoire (Dept. of Surgery, Ctr. Leon Berard, Lyon, France), and Jean-Yves Chapelon (LabTAU - U1032, INSERM, Lyon, France)

The objective of this clinical study was to validate the effectiveness, accuracy, tolerance, and safety of a HIFU treatment developed for the treatment of liver metastases. Fifteen patients were included. The transducer has a toroidal shape (diameter: 70 mm, radius of curvature: 70 mm) and was divided into 256 ultrasound emitters operating at 3 MHz. A 7.5 MHz ultrasound imaging probe was placed in the center of the device. All HIFU ablations were obtained in 40 s. The demarcation between ablated and non-ablated tissue was clearly apparent in ultrasound images and histology. In phase I (6 patients), we demonstrated that the dimensions of HIFU ablations measured on ultrasound imaging were correlated ($r = 0.88$) with dimensions measured during histological analysis. The average dimensions obtained from each HIFU ablation were a diameter of 21.0 ± 3.9 mm and a depth of 27.5 ± 6.0 mm. In phase II (9 patients), the HIFU ablations were centered on a target previously identified with a precision of 1–2 mm. It was demonstrated that HIFU ablations can be precisely located at 7.0 ± 2.3 mm from the target (expected distance 7.5 mm). This toroidal HIFU transducer achieved fast, selective, safe, and well-tolerated large volume liver ablation.

10:00

3aBAb3. Non-invasive toroidal high intensity focused ultrasound transducer for increasing the coagulated volume in depth. Jeremy Vincenot, David Melodelima, Francoise Chavrier, and Jean-Yves Chapelon (LabTAU - U1032, INSERM, 151 cours Albert Thomas, Lyon, France, jeremy.vincenot@inserm.fr)

A device composed of 32 elements (78 mm^2 each) arranged on a toroidal transducer (operating frequency: 2.5 MHz) was developed to increase the coagulated volume. To date, our previous work on toroidal transducers used the outer envelope of a torus as a reference surface. Here, the transducer geometry was based on the interior part of a torus. This produces a focus that is ring-shaped but the ultrasound beams also intersect between the principal focal ring and the transducer surface to form a secondary focal zone, which contributes to increase the size of the lesion. The radius of curvature was 70 mm with a diameter of 67 mm. A 7.5 MHz ultrasound imaging probe was placed in the center of the device. Twenty ablations were produced *in vitro* by using electronic beam steering, each ablation was created in 55 s. The average depth of intervening tissues (skin-fat-muscle) was 11 ± 1 mm and the average depth of liver tissues that have been spared was 21 ± 4 mm. No significant temperature rise in intervening tissues was measured (maximal temperature: 41°C). The dimensions of these ablations were an average diameter of 10 ± 1 mm and an average depth of 27 ± 3 mm.

10:20

3aBAb4. Improving the acousto-optic detection of high-intensity focused ultrasound lesions. Matthew T. Adams, Qi Wang (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, adamsm2@bu.edu), Robin O. Cleveland (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), and Ronald A. Roy (Mech. Eng., Boston Univ., Boston, MA)

Real-time acousto-optic (AO) sensing has been shown to non-invasively detect changes in *ex vivo* tissue optical properties during high-intensity focused ultrasound (HIFU) exposures. Although proof-of-concept experiments have been successful, the underlying parameters and mechanisms affecting the AO detectability of HIFU lesion formation are not well understood. In this work, a modeling based approach is used to improve the AO sensing of lesion formation during HIFU therapy. The angular spectrum method is used to model the acoustic field from the HIFU source and the temperature field, due to the absorption of ultrasound, is modeled using a finite-difference time-domain solution to the Pennes bioheat equation. Wavelength specific changes in tissue optical properties are calculated using a thermal dose model, calibrated by experimental data. The diffuse optical field is modeled using an open-source graphics processing unit accelerated Monte Carlo algorithm. The Monte Carlo algorithm is modified to account for light-sound interactions, using the acoustic field from the angular spectrum method, and to account for AO signal detection. Results will

demonstrate the important roles of optical wavelength selection, and illumination and detection configurations on the detectability of HIFU lesions by optical and AO sensing methods. [Work supported in part by NSF.]

10:40

3aBAb5. Determination of tissue injury thresholds from ultrasound in a porcine kidney model. Yak-Nam Wang, Julianna C. Simon, Bryan W. Cunitz, Frank L. Starr, Marla Paun (APL, CIMU, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, ynwang@u.washington.edu), Liggitt Denny (Dept. of Comparative Medicine, Univ. of Washington, Seattle, WA), Andrew P. Evan, James A. McAteer, James C. Williams (Dept. of Anatomy and Cell Biology, Indiana Univ., Indianapolis, IN), Ziyue Liu (Dept. of Biostatistics, Indiana Univ., Indianapolis, IN), Peter J. Kaczkowski (APL, CIMU, Univ. of Washington, Seattle, WA), Ryan S. Hsi, Mathew D. Sorensen, Jonathan D. Harper (Dept. of Urology, Univ. of Washington, Seattle, WA), and Michael R. Bailey (APL, CIMU, Univ. of Washington, Seattle, WA)

Therapeutic ultrasound has an increasing number of applications in urology, including shockwave lithotripsy, stone propulsion, tissue ablation, and hemostasis. However, the threshold of renal injury using ultrasound is unknown. The goal of this study was to determine kidney injury thresholds for a range of intensities between diagnostic and ablative therapeutic ultrasound. A 2 MHz annular array generating spatial peak pulse average intensities (I_{SPPA}) up to 30,000 W/cm^2 in water was placed on the surface of *in vivo* porcine kidneys and focused on the adjacent parenchyma. Treatments consisted of pulses of 100 μs duration triggered every 3 ms for 10 min at various intensities. The perfusion-fixed tissue was scored by three blinded independent experts. Above a threshold of 20,000 W/cm^2, the majority of injury observed included emulsification, necrosis, and hemorrhage. Below this threshold, almost all injury presented as focal cell and tubular swelling and/or degeneration. These findings provide evidence for a wide range of potentially therapeutic ultrasound intensities that has a low probability of causing injury. While this study did not examine all combinations of treatment parameters of therapeutic ultrasound, tissue injury appears dose-dependent. [Work supported by NIH DK43881, DK092197, and NSBRI through NASA NCC 9-58.]

11:00

3aBAb6. The origins of nonlinear enhancement in *ex vivo* tissue during high intensity focused ultrasound ablation. Edward Jackson, Robin O. Cleveland, and Constantin C. Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Old Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, edward.jackson@magd.ox.ac.uk)

Thermal ablation by high intensity focused ultrasound (HIFU) is an emerging technique for non-invasive treatment of tumors. One barrier to its wider use is that current imaging modalities are not able to directly monitor changes in the tissue during treatment. Nonlinear imaging has been suggested as a possible mode, but it is unclear whether the nonlinear enhancement is due to changes in the properties of treated tissue or cavitation. This study uses a finite-amplitude insertion technique to measure B/A of *ex-vivo* bovine liver as a function of temperature. The technique creates quasi-plane wave conditions by measuring the waveform in the near field of a 10 cm diameter 1 MHz unfocused source. The linear acoustic properties are measured in the same location and then B/A determined by fitting data to a numerical solution of the Burgers equation. Measurements are taken during heating in a water bath (at a slow rate) and HIFU exposure (at a fast rate). Cavitation is monitored during the HIFU exposures. Previous data suggested B/A doubles after heating. However, in these experiments, the changes were less than 20%. These data suggest that cavitation effects dominate changes in B/A.

WEDNESDAY MORNING, 5 JUNE 2013 512AE, 8:55 A.M. TO 12:00 NOON

Session 3aEA

Engineering Acoustics: Computational Methods in Transducer Design, Modeling, Simulation, and Optimization II

Daniel M. Warren, Chair
Knowles Electronics, 1151 Maplewood Dr., Itasca, IL 60134

Chair's Introduction—8:55

Invited Papers

9:00

3aEA1. Electret microphone modeling and optimization by combined finite element analysis and lumped-element techniques. David Schafer (Knowles Electron., 1151 Maplewood Dr., Itasca, IL 60143, david.schafer@knowles.com)

Electret microphone performance modeling must represent detailed coupled mechanical, electrostatic, and fluid dynamics of the moving diaphragm, stationary back electrode and intervening air layer (the "motor") while also evaluating the net performance characteristics which result. We model electret microphone response and noise by combined FEA and lumped-element methods. In detail, the diaphragm is treated as a tensioned membrane. Electrostatics are evaluated in locally parallel-plate fashion. Air layer dynamics are treated by reducing the exact thermoacoustic parallel-plate 3D squeeze solution to a 2D "transmission sheet" differential equation. Analyses (COMSOL FEA 2D PDE-mode) include initial diaphragm deflection upon back electrode polarization, then the harmonic responses to changes in frontside pressure, backside pressure, and polarization. Motor responses (frontside/backside volume flows and output current) are reduced to impedance and transfer characteristics of a three-port (front and back acoustic, electrical) element, which is then embedded in a traditional host lumped-element equivalent circuit (including amplifier). FEA-based motor characteristics over an array of frequencies lead to microphone frequency response and noise spectrum. Performance optimization is done by scripting motor geometry generation and FEA analysis as a subroutine, defining a space of adjusted parameters (motor dimensions, etc.) and running recurring FEA analyses generated by a downhill simplex algorithm.

9:20

3aEA2. Transducer modeling for optimal *in-situ* performance in a hearing instrument context. Martin Kuster (Sci. & Technol., Phonak AG, Laubisrütistr. 28, Stäfa 8712, Switzerland, kuster_martin@hotmail.com)

In a hearing instrument context, the transducers design has to be optimized for optimal performance in the installed configuration. This is typically behind or inside the outer ear of the instrument's wearer. Several examples will be shown where the *in-situ* acoustics is taken into account in the design or selection process of transducers. In this process, acoustic modeling is used extensively. Transducers are modeled by network analogs and they are coupled under defined impedance conditions to 3D numerical acoustics models, which incorporate the *in-situ* influence of, e.g., the wearer's ear anatomy. Moreover, in some cases, the modeling also includes part of the signal processing present in the hearing instrument.

Contributed Papers

9:40

3aEA3. Investigation of vibrations of piezoelectric spherical shells with axisymmetric holes. David A. Brown and Colton T. Brown (BTech Acoust., Adv. Tech. & Manuf. Cntr./ECE, UMass Dartmouth, 151 Martine St., Fall River, MA 02723, DBrown@BTechAcoustics.com)

An experimental investigation of the vibration of radially polarized thin-walled piezoelectric ceramic spherical shells with axisymmetric holes is presented. Piezoelectric spherical electroacoustic transducers having holes at their poles to permit passage of wires, cables, or structural members is of particular interest in underwater acoustics. The coupled vibrations follow three resonance branches corresponding to (0) azimuthal extensional modes, (1) bending flexural modes, and (2) meridional extensional modes. The resonances and effective electromechanical coupling coefficient for each mode as a function of the hole-to-sphere diameter ratio have been determined from admittance measurements. In the limit that the hole-to-sphere diameter ratio approaches zero, the (0) mode is dominant corresponding to the spherically symmetric breathing mode having a measured coupling coefficient of 0.546, which is consistent with the planar coupling coefficient for PZT-4 (Type I) material. In the limit that ratio approaches unity, the element in nearly cylindrical and the lowest (0) mode corresponds to the breathing mode of a ring having a coupling coefficient of about 0.33, which is consistent with the transverse (31) coupling coefficient for the material. The focus of the study is on determining the resonances and electromechanical coupling in the intermediate region and obtaining corresponding vibration mode shapes with a non-contact optical fiber displacement sensor.

10:00

3aEA4. Design of resonant frequencies of the piezoelectric actuator with integrated components. Jun Kuroda, Yasuharu Onishi, Motoyoshi Komoda (Common Platform Development Div., NEC CASIO Mobile Commun., LTD., 1753, Shimonumabe, Nakahara-Ku, Kawasaki, Kanagawa, Kanagawa 211-8666, Japan, j-kuroda@bq.jp.nec.com), Yasuhiro Oikawa, and Yoshio Yamasaki (Dept. of Intermedia Art and Sci., Waseda Univ., Tokyo, Japan)

The piezoelectric actuators have been applied to various electrical devices such as piezoelectric speakers, buzzers, haptics, and ultrasonic transducers. The improvement of the electromechanical conversion efficiency is the most important issue of piezoelectric actuator systems in mobile devices. The electrical power consumption of actuators must be suppressed as possible, due to mobile devices having small batteries. The frequency response around the mechanical resonance should be carefully designed for low power consumption driving. The resonant frequencies of the piezoelectric actuators consist of integrated components, for example, metal corns of ultrasonic speakers, are decided by the energy dispersion of the total system. Therefore, design factors such as sizes and physical properties of each component, are necessary to optimize the resonant frequencies for practical applications. The total energy of the piezoelectric system is described by Lagrange-Maxwell equation. Even though it is not easy to solve the differential equations written in the Lagrangian coordinate system by the exact calculation, useful information for design of the system can be derived by the approximate calculation. In this paper, we will indicate the design guideline to optimize resonant frequencies of the piezoelectric actuators with integrated components, based on the analysis in the Lagrangian coordinate system.

10:20

3aEA5. Calculation of characteristics of nonlinear normal waves in plates of lithium niobate for the designing of acousto-electronic devices. Alina Kuslyva and Valery Storozhev (Donetsk National Univ., bul. Kramatorskij 21/33, Kramatorsk, Donetsk 84331, Ukraine, kuslivaya@gmail.com)

The research of anharmonic effects is essential for the design of nonlinear acousto-electronic devices. Such effects involve the generation of nonlinear second harmonics in propagation of normal electroelastic waves in crystal plates. Thereby the analytical and numerical technique of the analysis of small nonlinear anharmonic effects in distribution of normal electroelastic waves in the layer of a trigonal piezocrystal of lithium niobate with thin short-circuited electroconductive coverings of sides has been developed. The research is based on the model of physically and geometrically nonlinear electroelastic deformation with finite deformations and Gibbs's function that includes quadratic and cubic components on deformations and characteristics of intensity of quasistatic electric field. The analysis of nonlinear wave effects is build on the representation of characteristics of a normal electroelastic wave in the form of the sum of summands, which are proportional to the powers of the small parameter. The analytical form has been received for the representations of functions of the elastic displacements, intensity, induction of quasistatic electric field in nonlinear second harmonics for the studied waves from the different modes of the dispersive spectrum. Quantitative estimates have been researched for the amplitude levels of second harmonics for normal electroelastic waves with variable frequencies.

10:40

3aEA6. A numerical study of non-collinear mixing of three-dimensional nonlinear waves in an elastic half-space. Zhenghao Sun, Hongguang Li, and Fucai Li (State Key Lab. of Mech. System and Vib., School of Mech. Eng.,Shanghai Jiao Tong Univ., Rm. B322, 800 Dongchuan Rd., Shanghai 200240, China, sunzhh@163.com)

Interactions of two non-collinear nonlinear ultrasonic waves in an elastic half-space with quadratic and hysteretic nonlinearity are investigated in this paper. The numerical problem is solved by a semi-discrete central scheme as well as a finite element method in two and three dimensions, respectively. Features and intensity distribution of the resonant wave are analyzed both in time and frequency domains, and the method of non-collinear wave mixing is proved to be a promising method, which is both effective and sensitive in material characterization and structure damage detection.

11:00

3aEA7. Simulative measures for structure borne sound radiation of composites. Matthias Klaerner (Inst. of Lightweight Structures, Chemnitz Univ. of Technol., Reichenhainer Str. 70, Chemnitz 09126, Germany, matthias.klaerner@mb.tu-chemnitz.de), Steffen Marburg (Insitute of Mech., Universtitaet der Bundeswehr München, Neubiberg, Germany), and Lothar Kroll (Inst. of Lightweight Structures, Chemnitz Univ. of Technol., Chemnitz, Germany)

Due to the high stiffness-to-weight ratio, composite structures tend to be acoustic sensitive. The sound emission of such radiating components is commonly measured by the sound power requiring the determination of the sound intensity in normal direction and—in numerical simulations—the sound pressure on the radiating surface. Assuming a unit radiation efficiency all-over the surface and neglecting local effects, the equivalent radiated power (ERP) is a common approach for an upper bound of structure borne noise. Therein, the sound power finally results from the squared velocity integrated over the radiating surface and the fluid impedance. As ERP usually requires extra post processing to consider the velocity in normal surface direction, the kinetic energy is essential in common FEA results including all velocity components apart from the normal direction, too. Moreover, ERP necessitates the knowledge of the radiating surfaces increasing the effort especially for complex geometries. Thus, the possibilities and limits of estimating the emitted sound power by the kinetic energy as well as using the ERP method will be shown. Test cases are a rectangular plate and a thin-walled bonded part with linear anisotropic material properties.

11:20

3aEA8. Transduction as energy conversion: Harvesting of acoustic energy in hydraulic systems. Ellen A. Skow, Kenneth Cunefare, and Alper Erturk (School of Mech. Eng., Georgia Inst. of Technol., Georgia Tech, Atlanta, GA 30332-0405, eskow3@gatech.edu)

Energy harvesting from acoustic energy sources is a form of transduction. While energy densities in typical airborne acoustic noise fields are extremely low, those in hydraulic systems may be orders of magnitude greater and represent an opportunity for direct energy conversion from piezoelectric materials for powering sensor and communication nodes. Hydraulic systems are challenging from a design perspective in that the device must be capable of withstanding static pressures up to and exceeding 35 MPa, while being simultaneously exposed to dynamic pressures on the order of 3.5 MPa. Hydraulic pressure energy harvester devices have been developed to exploit the high energy densities of dynamic pressures in hydraulic systems. There is an immediate application for this technology in that state-of-the-art hydraulic hose and piping systems employ integral sensor nodes for structural health monitoring for early detection of incipient failures. This paper presents the acoustic and electromechanical modeling of the piezoelectric power output from dynamic pressure in terms of the force transmitted into an energy harvester designed for hydraulic systems.

11:40

3aEA9. Accurate determination of piezoelectric ceramic constants using a broadband approach. Nicolas Perez (Centro Universitario de Paysandu, Universidad de la Republica, Montevideo, Uruguay), Marco Aurelio B. Andrade, Ronny C. Carbonari (Biomedical Eng., Federal Univ. of ABC, Sao Paulo, Brazil), Julio C. Adamowski (Mechatronics Eng., Univ. of Sao Paulo, Sao Paulo, Brazil), and Flavio Buiochi (Mechatronics Eng., Univ. of Sao Paulo, Av. Prof. Mello Moraes, 2231, Butanta, Sao Paulo, SP 05508-030, Brazil, fbuiochi@usp.br)

Piezoceramic property values are required for modeling piezoelectric transducers. Most datasheets present large variations in such values. For precise simulations, adjustments are necessary. Recently, the authors presented a methodology to obtain the real part of ten material constants of piezoelectric disks. It comprises four steps: experimental measurements, identification of vibration modes and their sensitivity to material constants, preliminary identification algorithm, and final refinement of the constants using an optimization algorithm. Given an experimental electrical impedance curve of a piezoceramic and a first estimate for the material constants, the objective is to find the constants that minimize the difference between the experimental and numerical curves. Using a new finite element method routine implemented in MATLAB, the original methodology was extended to obtain the corresponding imaginary part of all the material constants. Results of sensitivity analysis for the imaginary part and the guidelines to construct an algorithm are presented. This complex model allows adjusting the amplitude over a wide frequency range, as opposed to the models described in the literature. It is applied to 1-MHz APC850 disks with diameters of 10 and 20 mm. The methodology was validated by comparing the numerical displacement profiles with the displacements measured by a laser Doppler vibrometer.

WEDNESDAY MORNING, 5 JUNE 2013 — 510C, 8:55 A.M. TO 12:20 P.M.

Session 3aED

Education in Acoustics and Psychological and Physiological Acoustics: Learning by Listening: Education in Acoustics Based on Listening

Akira Nishimura, Cochair
Media and Cultural Studies, Tokyo Univ. of Information Sci., 4-1, Onaridai, Wakaba-ku, Chiba 2658501, Japan

Kaoru Ashihara, Cochair
AIST, AIST Tsukuba Central, 1-1-1 Higashi, Tsukuba 3058566, Japan

Chair's Introduction—8:55

Invited Papers

9:00

3aED1. Technical ear training: Tools and practical methods. Jason Corey (School of Music, Theatre & Dance, Univ. of Michigan, 1100 Baits Dr., Ann Arbor, MI 48109-2085, coreyja@umich.edu)

Broadly defined, technical ear training seeks to make associations between aural impressions of sound quality and quantifiable characteristics of audio signal processing and acoustical measurements. Technical ear training typically focuses on attributes of sound such as spectral balance (e.g., filtering and parametric equalization); dynamic range of musical signals (including artifacts produced by dynamics processing); reverberation, delay, and early reflections (from real acoustic spaces or generated artificially); and spatial extent (width and depth). These elements of recorded sound can be broken down into graduated levels of audibility for the development of critical listening skills. With repeated and regular practice of carefully chosen exercises, listeners can gain increased sensitivity to subtle details of sound, as well as efficiency and accuracy in identifying specific parameters of signal processing by ear. With applications

primarily in sound recording and production, technical ear training is also highly relevant to the evaluation of acoustical spaces as a complement to objective measurements. This presentation will review a selection of software modules developed by the author to teach critical listening skills to undergraduate students. The author will also discuss some practical methods and exercises used for teaching technical ear training and critical listening.

9:20

3aED2. Learning acoustic phonetics by listening, seeing, and touching. Takayuki Arai (Dept. of Information and Commun. Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, arai@sophia.ac.jp)

There is a huge volume of written textbooks available in virtually every modern field, including acoustic phonetics. However, in areas dealing with acoustics, learners often face problems and limitations when they deal with only written material and no audio or visual information. As one response to this problem, we have developed several sets of physical models of the human vocal tract and have shown that they are extremely useful for intuitive understanding. In addition, we also developed a tool called "Digital Pattern Playback." Another solution is an online version featuring demonstrations. We are currently collecting materials, mainly in the form of sounds, for educational purposes in acoustics and phonetics and are releasing them as "Acoustic-Phonetics Demonstrations" through our Web site. These demonstrations are designed for students in linguistics, phonetics and phonology, speech pathology, audiology, psychoacoustics, speech engineering, and others. However, potential users are not limited to these groups, as we feel that a wide range of learners can obtain tremendous benefits from the demonstrations, including those who are studying foreign languages or patients undergoing speech articulation therapy. [Work partially supported by a Grant-in-Aid for Scientific Research (24501063) from the Japan Society for the Promotion of Science.]

9:40

3aED3. Enriching the aural experience in audio education. Alex Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

Audio education's essential outcome is aural. Lectures and readings on aesthetics, techniques, and technologies can never communicate audio concepts effectively without critical elucidation through sound. Quality audio education has always made frequent use of laboratories, recording sessions, and critical listening classrooms to keep sound at the center of student learning. Recently authored and published web-based multimedia and digital audio workstation self-study experiences are discussed and demonstrated. The sonic illustrations, visual reinforcement, and associated interactivity are found to provide meaningful pedagogical advancements in audio education.

10:00

3aED4. Sonority in British English. Yoshitaka Nakajima, Kazuo Ueda (Dept. of Human Sci., Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka, Fukuoka 815-8540, Japan, nakajima@design.kyushu-u.ac.jp), Shota Fujimaru (Human Sci. Course, Kyushu Univ., Fukuoka, Japan), and Yuki Ohsaka (The 21st Century Program, Kyushu Univ., Fukuoka, Japan)

Our previous study on British English speech [Nakajima *et al.*; Fechner Day (2012), Ottawa] was extended into the domain of phonology. Factor analyses were performed on the power fluctuations of the outputs of 19 critical-band filters, which separated British English sentences uttered by two female speakers and one male speaker into narrow-band signals. A database of British English spoken sentences [The ATR British English speech database (Campbell, 1993)] was used for the present analysis, because each identifiable speech sound was indicated in the speech waveform with a label of a phonetic symbol. About 80% of 31,663 labels were considered to represent English phonemes. Three factors appeared as in our previous research [Ueda *et al.*; Fechner Day (2010), Padova], and one of them corresponding to a frequency range of about 600–1800 Hz was closely related to sonority or aperture described in linguistics literature. The acoustic sonority could be related to a few phonological phenomena: (1) A sonorant consonant immediately after an obstruent can be a syllable nucleus, (2) a consonant cluster at the beginning of a word mostly begins with an obstruent, and (3) a short schwa cannot be a nucleus of a stressed syllable.

Contributed Papers

10:20

3aED5. Toward the development of objective difficulty measure in technical ear training tasks. Atsushi Marui and Toru Kamekawa (Faculty of Music, Tokyo Univ. of the Arts, 1-25-1 Senju, Adachi, Tokyo 120-0034, Japan, marui@ms.geidai.ac.jp)

Technical Ear Training is a method to improve the ability to focus on a specific sound attribute. It is also used to be able to communicate using the common language used in the industry such as Hz and dB. Although it is essential to gradually harden the task difficulty for successful technical ear training, the objective measure of the difficulty is still not known. Therefore, the tasks are decided by the teacher's own ears and experiences, leading to inefficiency when students want to train themselves in the teacher's absence. As the first step toward understanding this tacit ability of knowing the task difficulty, the authors investigated the correlation between the students' subjective ratings of the task difficulty and the physical measures calculated from the sound materials used in the training. A linear regression model ($R^2 = 0.629$) which predicts the subjective task difficulty from residual of the linear fit through the spectra of the sound material was created. This model may provide a firm step toward the goal of developing objective difficulty measure in technical ear training.

10:40

3aED6. Human echolocation system using a miniature dummy head. Shunsuke Uchibori, Masataka Kinoshita (Faculty of Life and Med. Sci., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, dmm1028@mail4.doshisha.ac.jp), Kaoru Ashihara (National Inst. of Adv. Industrial Sci. and Technol., Ibaraki, Japan), Tetsuo Ohta, and Shizuko Hiryu (Faculty of Life and Med. Sci., Doshisha Univ., Kyotanabe, Japan)

To promote understanding of sonar mechanisms in bats, we propose a novel tool that makes echolocation available for humans. In this method, ultrasonic echoes are captured by a miniature dummy head so that they can be converted to binaural audible sounds using time expansion.In order to examine the effectiveness of this technique, perceptual listening tests were conducted on human listeners with normal hearing.The sounds (white noise with frequencies between 5 and 90 kHz with 0.7-s duration, including 0.05-s rise/fall time) were recorded at a distance of 1 m from a loudspeaker

using two condenser microphones that were placed in the ear canals of a 1/7 size miniature dummy head. The recorded ultrasounds were 1/7-times pitch converted, and then were presented to the listener through headphones. As a result, the listeners perceived correct directions of the pitch converted sounds, which were recorded using the miniature dummy head, although front-back error was occasionally observed. When the miniature dummy head was rotated during the recording, the listeners perceived the movement of the sounds as out-of-head sound localization. The miniature dummy head may provide humans with a tool to understand biosonar mechanisms.

11:00

3aED7. Capturing spatial audio information by using a miniature head simulator. Kousuke Taki (Tokyo City Univ., 105, 2-4-13, Noge, Setagaya-ku, Tokyo 158-0092, Japan, g0922040@tcu.ac.jp), Kaoru Ashihara (National Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Ibaraki, Japan), and Shogo Kiryu (Tokyo City Univ., Tokyo, Tokyo, Japan)

A conventional dummy head or a microphone array with a number of microphones has been used to record spatial audio information. We propose an audio capture system called "Miniature head simulator" that consists of a microphone system and a signal processor. Instead of using a conventional dummy head, a small microphone system that consists of three omni-directional microphones is used to capture acoustic signals. The signals are then encoded to a custom data format and transmitted online. The transmitted data can be decoded and reproduced as binaural signals. Because of its size and weight, the conventional dummy head has been used exclusively for the research purpose. A miniature head simulator can provide much more convenient tools to record spatial audio information. Since it deals with only three channels of audio stream, data can be processed with relatively low computational cost. By using a miniature head simulator, spatial audio information can be streamed to the browser or even to the smartphones of the end-users. It can be used in a remote acoustic sensor and remote control system, sound detection system, video conference, and various other fields of industry.

11:20

3aED8. Effectiveness of technical listening training in Department of Acoustic Design of Kyushu University. Kazuhiko Kawahara, Masayuki Takada, and Shin-ichiro Iwamiya (Faculty of Design, Kyushu Univ., 4-9-1 shiobaru, Minami-ku, Fukuoka 815-8540, Japan, kawahara@design.kyushu)

What is the professional listening? The listening ability of Sound/Acoustic Professionals listening is categorized into three phases: the ability to discriminate between different sounds, the ability to correlate the auditory difference with the physical properties of sounds, and the ability to imagine the proper sounds when given the acoustic properties of the sounds. These kinds of ability can be trained through listening training. Furthermore, through trainings, trainees can share same auditory experiences. The shared experiences improve ability of trainees to express their auditory impression with appropriate words and this ability contributes to smooth communication on auditory imagery in their group. In this paper, as a listening training, the overview and effectiveness of Technical Listening Training in Kyushu University is described. To evaluate the effectiveness of training, the average correct answer ratios of trainees were examined through the training. The improvement of correct answer ratios was observed. We could show the effectiveness of Technical Listening Training.

11:40

3aED9. Effect of speaking rate variation on the perception of singleton and geminate consonants in Japanese by native and Korean listeners. Mee Sonu, Takayuki Arai (Faculty of Sci. and Technol., Sophia Univ., 7-1 Kioi-Cho, Chiyoda-ku, Tokyo 102-8554, Japan, sonumeephonetic@gmail.com), Hiroaki Kato (National Inst. of Information and Commun. Technol., Kyoto, Japan), and Keiichi Tajima (Dept. of Psych., Hosei Univ., Tokyo, Japan)

Perception of phonemic length contrasts in Japanese is difficult for non-native listeners. To better understand the source of this difficulty, the present study investigated native Korean listeners' perception of consonant length contrasts at different speaking rates. Stimuli were created by modifying the duration of the second consonant of a non-word /ereC:e/ along a continuum to /ereCe/, where C was /k/ or /s/. The base words were spoken by a professionally trained native Japanese speaker with a carrier sentence at three rates, fast, normal, slow. Twenty-seven native Korean and eleven native Japanese listeners participated in a perception test. They listened to one of the created stimuli and identified whether the second consonant was singleton or geminate. Results show that even though Korean listeners' perceptual boundary location between singleton and geminate consonants shifted according to speaking rate in a similar manner as the natives, their boundary location was more variable than native listeners at all speaking rates. Korean listeners also showed greater perceptual boundary width than Japanese listeners. These results suggest that Korean listeners have ambiguous criteria for phonemic length contrasts. Results are discussed in terms of the perceptual similarity between Korean and Japanese consonants. [Work supported by JSPS.]

12:00–12:20 Demonstrations

Session 3aMU

Musical Acoustics and Signal Processing in Acoustics: Aeroacoustics of Wind Instruments and Human Voice II

Shigeru Yoshikawa, Cochair
Grad. School of Design, Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan

Xavier Pelorson, Cochair
CNRS, 11 rue des mathematiques, Saint Martin d'Heres 38330, France

Invited Papers

9:00

3aMU1. Adaptive stabilized finite element framework for simulation of vocal fold turbulent fluid-structure interaction and towards aeroacoustics. Johan Jansson (KTH Royal Inst. of Technol. and Basque Ctr. for Appl. Mathematics, CSC KTH, Stockholm SE-10044, Sweden, jjan@kth.se)

As a step toward building a more complete model of voice production mechanics, we assess the feasibility of a fluid-structure simulation of the vocal fold mechanics in the Unicorn incompressible Unified Continuum framework. The Unicorn framework consists of conservation equations for mass and momentum, a phase function selecting solid or fluid constitutive laws, a convection equation for the phase function and moving mesh methods for tracking the interface, and discretization through an adaptive stabilized finite element method. The framework has been validated for turbulent flow for both low and high Reynolds numbers and has the following features: implicit turbulence modeling (turbulent dissipation only occurs through numerical stabilization), goal-oriented mesh adaptivity, strong, implicit fluid-structure coupling, and good scaling on massively parallel computers. We have applied the framework for turbulent fluid-structure interaction simulation of vocal folds, and present recent results. Initial steps toward aeroacoustics have been carried out in the framework (exhaust system and landing gear applications), and will be presented as well as the current state of aeroacoustic modeling for the human voice in the framework.

9:20

3aMU2. Synergistic interactions underlying the production of voice. Tokihiko Kaburagi (Grad. School of Design, Kyushu Univ., Shiobaru 4-9-1, Minami-ku, Fukuoka 815-8540, Japan, kabu@design.kyushu-u.ac.jp)

This paper presents recent progress in research examining the mechanisms of voice generation and a method for physiologically based speech synthesis. The overarching goal of this research is the precise modeling of interactions among physical systems involved in the processes underlying voice generation. In the basic voice generation system, a flow-structure interaction between glottal flow and the vocal folds causes self-oscillations of the folds, where flow separation, a nonlinear aerodynamic phenomenon, plays an important role. The fluid dynamic theory implies that a thin boundary layer formed near the glottal wall characterizes the flow behavior, including flow separation, jet formation, and pressure loss across the channel. We therefore use the interactive boundary layer method to analyze glottal flow and show how the flow-structure interaction is effective in maintaining vocal fold oscillations. In addition, the interaction between the voice generation system and the vocal-tract filter, i.e., the source-filter coupling, has been found to involve nonlinear factors in speech, such as skewing of glottal flow, unsteadiness in vocal fold oscillations, and transitions in voice register.

9:40

3aMU3. Computational analysis of the dynamic flow in single-reed woodwind instruments. Andrey R. da Silva (Structures and Civil Eng., Federal Univ. of Santa Maria, Av. Roraima 1000, Santa Maria, Rio Grande do Sul 97050421, Brazil, andrey@eac.ufsm.br), Shi Yong, and Gary Scavone (Music, McGill Univ., Montreal, QC, Canada)

The dynamics of the air flow within the mouthpiece of single-reed wind instruments makes an important contribution to the acoustic behavior of this type of system, particularly during transient regimes. In this work, a two-dimensional numerical model of the mouthpiece-reed system is used to evaluate the behavior of the flow within the mouthpiece at three different frequencies within the playing range of the clarinet. The relationship between volume flow and blowing pressure, as well as the behavior of the vena contracta during one duty cycle are compared with the available analytical models and with recent experimental observations.

10:00

3aMU4. Theoretical and experimental study of glottal geometry in phonation. Xavier Pelorson, Annemie Van Hirtum, Bo Wu, and Fabrice Silva (Depertement Parole et Cognition, Gipsa-Lab, 11 rue des mathematiques, Saint Martin d'Heres 38330, France, xavier.pelorson@gipsa-lab.grenoble-inp.fr)

Most existing theoretical models of phonation assume that the vocal folds are parallel and that the glottis forms a two-dimensional channel for the flow. However, during phonation the vocal folds can be very accurately abducted or adducted using intrinsic muscles acting on the arytenoid cartilages. The resulting shape of the glottis can then vary between an almost uniform slit (when the vocal folds are parallel) to a V shape

(with an angle between the vocal folds up to 20 °). Further, the vocal folds surface can present some irregularity which can be severe, in some pathological cases such as cysts or nodules. In this paper, we present a theoretical and experimental study performed in order to evaluate and to predict these geometrical effects. Several theoretical models to predict the pressure losses in a non-uniform glottis will be presented and tested against an *in-vitro* experimental set-up using a self-oscillating latex replica of the vocal folds. The angle between the artificial folds could be controlled using micrometers while surface irregularities could be simulated by inserting small spheres with various masses and diameters.

10:20

3aMU5. Influence of epithelium and fiber locations on glottal closure and sound production at soft-phonation conditions. Zhaoyan Zhang and Yue Xuan (UCLA School of Med., 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Previous studies showed that isotropic vocal fold models often vibrated with incomplete glottal closure at onset despite the vocal folds were in contact at rest. This contrasts with human phonation in which complete glottal closure is observed even during soft phonation with minimal or low laryngeal muscle contraction. Based on previous experimental studies, we hypothesize that this difference in glottal closure patterns is due to the relatively large stiffness in the anterior-posterior direction or the presence of the epithelium layer. These hypotheses were tested in self-oscillating physical vocal-fold models, with anisotropic stiffness conditions simulated by fibers loosely imbedded at different locations in otherwise isotropic vocal folds. The results showed that, compared to isotropic one-layer models, the presence of a stiff epithelium layer led to complete glottal closure along the anterior-posterior direction, increased maximum glottal opening, strong excitation of high-order harmonics in the resulting voice spectra and reduced noise production. Similar improvement in glottal closure and high-order harmonics excitation was observed with fibers in the cover layer, but to a less degree. Presence of fibers in the body-layer led to reduced maximum glottal opening but did not yield noticeable improvement in glottal closure and harmonic excitation. [Work supported by NIH.]

Contributed Papers

10:40

3aMU6. Synchronous visualization of multimodal measurements on lips and glottis: Comparison between brass instruments and the human voice production system. Thomas Hézard (IRCAM - CNRS UMR 9912 - UPMC, 1, place Igor Stravinsky, Paris 75004, France, thomas.hezard@ircam.fr), Vincent Fréour (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., Montréal, QC, Canada), René Caussé, Thomas Hélie (IRCAM - CNRS UMR 9912 - UPMC, Paris, France), and Gary P. Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., Montréal, QC, Canada)

Brass instruments and the human voice production system are both composed of a vibrating "human valve" (constriction in a pipe) coupled to an acoustic resonator: lips coupled to the brass instrument or vocal folds coupled to the vocal tract. In both cases, the aeroacoustic coupling is responsible for the self-oscillations and the large variety of regimes. Additionally, brass instruments and voice share difficulties for the *in-vivo* measurement of the exciter activity. Hence, the development of a common tool is relevant. It is also relevant to explore the effect of some known differences between these systems, namely, the strength of the coupling and the physiological characteristics. This paper introduces components for the development of such a tool. First, two corpuses of multimodal measurements are presented: one for a singer's larynx during sustained vowels, one for a musician's lips during sustained notes. They include high-speed video (HSV) recordings, electrical impedance (EI) measurements (electrolabiograph/electroglottograph), and audio recordings (AR). Then, we introduce two estimation algorithms: (AW) one of the opening area waveforms from videos, (LF) one of the LF-model parameters on these waveforms. Moreover, we build a video tool displaying, synchronously, the signals (HSV), (AW), (EI), and (AR) in time and frequency domains. Finally, this tool is exploited to exhibit common behaviors and relevant differences between brass instruments and human voice.

11:00

3aMU7. Physical modeling of bilabial plosives production. Louis Delebecque, Xavier Pelorson, Denis Beautemps, and Xavier Laval (Speech and Cognition, GIPSA-lab, 11 rue des Mathématiques, Grenoble Campus BP46, Saint-martin d'hères F - 38402, France, louis.delebecque@gipsa-lab.grenoble-inp.fr)

The context of this study is the physical modeling of speech production. The first step of our approach is to realize *in vivo* measurements during the production of the vowel-consonant-vowel sequence /apa/. This measurements concerns intra oral pressure, acoustic pressure radiated at the lips and labial parameters (aperture and width of the lips) derived from a high-speed video recording of the subject's face. In a second time, theoretical models from speech production literature are under investigation to describe air flow in the lips. Their prediction are compared with measurements obtained using an experimental set-up including a replica of vocal folds, which are able to self-oscillate and a rigid replica of lips. Finally, this validation allows us to achieve numerical simulations of the sequence /apa/. The comparison between the measured intra oral pressure and the simulated one leads us to take into account for the cheeks expansion in the physical modeling of bilabial plosives.

11:20

3aMU8. Influence of cross section shape on the outcome of a two-mass model. Bo Wu, Annemie Van Hirtum (Gipsa-Lab, Grenoble Univ., 11 rue des Mathematiques, Grenoble 38402, France, annemie.vanhirtum@gipsa-lab.grenoble-inp.fr), and Xiaoyu Luo (Dep. of Mathematics, Glasgow Univ., Glasgow, United Kingdom)

For the last decades, the two-mass model has shown its value to model the fluid-structure interaction during voiced speech production. Its main interest lies in its simplicity since it allows a quasi-analytical solution for a complex phenomenon using only a limited amount of physical meaningful parameters. Nevertheless, the use of the two mass model with respect to model speech pathologies can be questioned due to the lack of detail in the used mechanical and/or flow model. In the current paper, we focus on the influence of the cross section shape taken into account. Indeed, varying the cross section shape is likely to alter the pressure distribution due to viscous effects. The influence of the cross section shape on the flow outcome is modeled as well as experimentally assessed. Next, the influence of varying the cross section shape on phonation parameters such of the threshold pressure and fundamental frequency are addressed by considering a linear stability analysis of the two mass model.

11:40

3aMU9. The spectral and acoustical impact of vowel changing in choral tuning. Bertrand Delvaux and David Howard (Dept. of Electron., Univ. of York, Heslington, York, York YO10 5DD, United Kingdom, bertrand.delvaux@gmail.com)

Mixed soft/solid models of the vocal tract were moulded with a 3D rapid prototyping technique based on MRI data obtained from two male singers during the phonation of five English vowels as in hard, stern, neap, port, and food. The replicas are used to assess the interaction of several vocal tracts in different settings: twice the same singer or two different singers, singing on the same vowel or on different vowels, on a consonant or a dissonant interval. The spectral output is analyzed and the acoustical output is submitted to a listening test to evaluate the spectral and acoustical grounds for an interval/chord to perceptually sound "in tune."

Session 3aNSa

Noise, ASA Committee on Standards, Engineering Acoustics, and Structural Acoustics and Vibration: Wind Turbine Noise I

Nancy Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton St., Boston, MA 02118

Paul Schomer, Cochair
Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821

Sheryl Grace, Cochair
Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215

Invited Papers

9:00

3aNSa1. Activities of the Acoustical Society of America's subcommittee on wind turbine noise and some studies being done. Nancy Timmerman (Nancy S. Timmerman, P.E., 25 Upton St., Boston, MA 02118, nstpe@hotmail.com)

This paper will document the activities of the Acoustical Society of America's (ASA's) subcommittee (of the Panel on Public Policy) on Wind Turbine Noise, including what technical committees are represented, what special sessions will be held in the future, and the goal to generate a policy statement on the topic. The author, who is Chair of this subcommittee, will also describe what other current studies are or have been done in Massachusetts (in the United States) and, if applicable, elsewhere.

9:20

3aNSa2. Development of a real time compliance system for wind farms regulated by ambient-relative noise standards. Michael Hankard (Hankard Environ., 211 East Verona Ave., Verona, WI 53593, mhankard@hankardinc.com)

Some noise level regulations in the United States require wind turbine farms to not exceed the ambient sound levels at nearby residences by more than a fixed amount. For the project discussed herein, compliance with such a regulation requires curtailment of turbine operations to some degree at times when turbine operations are at or near maximum, atmospheric conditions are conducive to sound propagation, and sound from other sources including vegetation rustle are at a minimum. Based on the analysis of months of time-synchronized sound, meteorological, and operations data, a system was developed to assess compliance on a real-time, ongoing basis. The primary element in the determination of compliance is the shape of the one-third octave band spectrum at the residence, augmented by ground and hub-height meteorological conditions, and wind farm operations information. Without the spectral filter, curtailment would need to take place under a broader array of meteorological conditions to ensure compliance, which would result in loss of power generation revenue. This paper will describe the data collection and analysis methods, the development of the spectral filter, and the results of field testing including both the partial and entire shut down of the wind farm.

9:40

3aNSa3. Criteria for wind-turbine noise immissions. George Hessler (Hessler Assoc., Inc., 3862 Clifton Manor Place, Ste. B, Haymarket, VA 20169, George@HesslerAssociates.com) and Paul Schomer (Schomer and Assoc., Inc., Champaign, IL)

Each of the two authors has developed recommended single, 24-h, constant wind turbine noise criterion; the criteria are constants because wind turbine noise is basically not adjustable. Hessler develops his criterion from his knowledge of how wind turbine noise is being regulated at the local, state, and national levels, from regulations in other countries, and from his extensive experience with numerous wind turbine projects. Schomer develops his recommended criterion on the basis of existing national and international standards; notably ISO 1996-Part 1 and ANSI/ASA S12.9 parts 4 and 5. Ultimately, Hessler comes up with a single, 24-h A-weighted average criterion of 40 dB, and Schomer comes up with a 24-h A-weighted average criterion of 39 dB. These two researchers have decidedly different backgrounds, different experience, and a slight difference in orientation towards the industry. Thus, it is remarkable that these two criteria, derived in such different ways result in nearly identical 24-h A-weighted criteria levels. Although there is essential agreement in immissions criterion, there are variables debated herein for both modeling wind turbine emissions and certifying such emissions at far-off receptors that could result in a 10 dBA difference in the actual immissions level.

10:00

3aNSa4. Prevalence of complaints related to wind turbine noise in northern New England. Kenneth Kaliski (RSG Inc., 55 Railroad Row, White River Junction, VT 05001, ken.kaliski@rsginc.com)

As of September 2012, there were a dozen large operating wind projects with a total capacity of approximately 565 MW in northern New England, with more coming online by the end of the year. This paper evaluates the prevalence of noise complaints to regulatory authorities from those wind projects. Where possible, the exposure of residences to wind turbine sound is calculated. Exposure is

estimated through standard ISO 9613-2 modeling procedures. A comparison of the exposure of complainants and non-complainants is made with the goal of assessing the prevalence of complaints at various modeled sound levels.

10:20–10:40 Break

10:40

3aNSa5. Can wind-turbine sound that is below the threshold of hearing be heard? Paul Schomer (Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

This paper is geared toward wind-turbine sound, but it is really a simple variation on the basic concepts that this author used in the development of loudness-level-weighted sound exposure (Schomer *et al*., J. Acoust. Soc. Am. **110**(5, Pt. 1), 2390–2397 (2001)] and of Rating Noise Curves (RNC) [Schomer, Noise Control Eng. J. **48**(3), 85–96 (2000)], which are used in our Standard, ANSI/ASA S12.2 Criteria for evaluating room noise. The fundamental issue is: Can we hear slowly surging or pulsating sounds for which the LEQ spectrum is below the threshold of hearing, where "slowly" means that the pulses come at a rate that is no faster than about 4 pulses per second? The short answer is yes, and the longer answer is that this effect is a function of the spectral content and becomes more-and-more prominent as the spectral content goes lower-and-lower in the audible frequency range. So surging or pulsing sound that is primarily in the 16 or 31 Hz octave bands will show the greatest effect. This paper shows the applicability of these results to wind-turbine sound. Variation in the threshold of hearing at low frequencies is an additional factor that also is discussed in this paper.

11:00

3aNSa6. Amplitude modulation of audible sounds by non-audible sounds: Understanding the effects of wind turbine noise. Jeffery Lichtenhan and Alec Salt (Otolaryngology, Washington Univ. in St. Louis, 660 South Euclid, St. Louis, MO 63110, LichtenhanJ@ent.wustl.edu)

Our research has suggested a number of mechanisms by which low-frequency noise could bother individuals living near wind turbines: causing endolymphatic hydrops, exciting subconscious pathways, and amplitude modulation of audible sounds. Here we focus on the latter mechanism, amplitude modulation. We measured single-auditory-nerve fiber responses to probe tones at their characteristic frequency in cats. A 50 Hz tone, which did not cause an increase in spontaneous firing rate (i.e., was not audible to the fiber when presented alone) was used to amplitude modulate responses to the probe tone. We found that as probe frequency decreased, a lower level of the low-frequency non-audible tone was needed to achieve criterion amplitude modulation. In other words, low-frequencies that are coded in the cochlear apex require less low-frequency sound pressure level to be amplitude modulated as compared to higher-frequencies that are coded in the cochlear base. This finding was validated, and extended to lower frequencies, by amplitude modulating gross measures of onset-synchronous (compound action potentials) and phase-synchronous (auditory nerve overlapped waveforms) in guinea pigs. Our results suggest that that infrasound generated by wind turbines may cause amplitude modulation of audible sounds, which is often the basis for complaints from those living near wind turbines.

11:20

3aNSa7. Generation of wind turbine noise signature for use in lab environment. Aleks Zosuls (Biomedical Eng., Boston Univ., Boston, MA), R. Morgan Kelley (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, sgrace@bu.edu), David Mountain (Biomedical Eng., Boston Univ., Boston, MA), and Sheryl Grace (Mech. Eng., Boston Univ., Boston, MA)

The fact that wind turbines produce infrasound continues to draw attention and discussion. Some argue that while the infrasound level produced by wind turbines is quite low, it still may be affecting the vestibular system or the hearing system, particularly via activation of the outer hair cells. Others hypothesize that the infrasound may be inducing whole body, chest cavity, or other human organ resonance. In order to study these hypotheses, it is first necessary to be able to recreate the turbine noise signature in a lab environment. Thus, the goal of this work is to create an acoustic system that can produce low-level infrasound. The system requirements include low cost, high fidelity, and imperceptible structural coupling to the lab. In addition, the system must be able to produce a broadband spectrum as well as a single tone. Progress toward the design of this audio system is discussed in this paper.

Contributed Papers

11:40

3aNSa8. Wind turbine sound prediction–The consequence of getting it wrong. William K. Palmer (TRI-LEA-EM, 76 Sideroad 33-34 Saugeen, RR 5, Paisley, ON N0G2N0, Canada, trileaem@bmts.com)

The application to permit a wind turbine power development usually involves submission of a prediction for the sound level that will occur at residences, schools, places of worship, and elsewhere people gather for restorative rest. This paper uses the example of a wind power development, and follows iterations taken to finalize the sound level prediction. The paper provides quantitative information collected since the start up of the wind power development on measured sound levels and octave band distribution; and qualitative observations on the special characteristics of the sound. Actual observations are compared to the predictions. More importantly, the paper reviews the consequences self-reported in qualitative interviews by citizens living with the changed environment after four years of operation of the wind power development. Reported impacts included difficulty sleeping, loss of jobs, and changes to social relationships, caregiving, pursuit of hobbies, leisure, learning, and overall health. Changes in measured health outcomes are identified. Both the quantitative and qualitative findings justify revision of the permitting process.

12:00

3aNSa9. Predicting underwater radiated noise levels due to the first offshore wind turbine installation in the United States. Huikwan Kim, James H. Miller, and Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Road, Narragansett, RI 02882, hkkim524@my.uri.edu)

Noise generated by offshore impact pile driving radiates into the air, water, and sediment. Predicting noise levels around the support structures at sea is required to estimate the effects of the noise on marine life. Based on high demands developing renewable energy source, the United States will

begin the first pile driving within one to two years. It is necessary to investigate acoustic impact using our previously verified coupled Finite Element (Commercial FE code Abaqus) and Monterey Miami Parabolic Equation (2D MMPE) models [J. Acoust. Soc. Am. **131**(4), 3392 (2012)]. In the present study, we developed a new coupled FE-MMPE model for the identification of zone of injury due to offshore impact pile driving. FE analysis produced acoustic pressure outputs on the surface of the pile, which are used as a starting field for a long range 2D MMPE propagation model. It calculates transmission loss for N different azimuthal directions as function of distance from the location of piling with the inputs of corresponding bathymetry and sediment properties. We will present predicted zone of injury by connecting N different distances of equivalent level fishes may get permanent injury due to the first offshore wind farm installation in the United States.

WEDNESDAY MORNING, 5 JUNE 2013 — 511CF, 9:00 A.M. TO 11:40 A.M.

Session 3aNSb

Noise: Aviation, Aviation Engines, and Flow Noise

Victor Sparrow, Chair
Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802

Contributed Papers

9:00

3aNSb1. A robust numerical approach for prediction of turbofan engine noise. Kaveh Habibi, Hao Gong, and Luc Mongeau (McGill Univ., Apt. 713, 3555 Berri St., Montreal, QC H2L4G4, Canada, kaveh.habibi@mail.mcgill.ca)

Noise from the aircraft jet engines is still the dominant source in takeoff condition. Government regulations of community noise are getting more stringent, creating significant challenges for aircraft manufacturers to meet these requirements. Expensive nature of experiments as well as accessing to powerful computers have provided new impetus for computational studies. In this paper, an efficient and robust numerical scheme, namely the Lattice Boltzmann Method was used to simulate the sound created by typical internal mixing nozzles with forced mixers. The simulation includes capturing the time-resolved flow characteristics and large scale turbulent structures. The sub grid scales were modeled using the renormalization group (RNG) forms of the standard k-ε equations. Several test cases including cold and hot core flow experiments conducted by NASA were selected for computational setup and validation purposes. The far field sound was predicted using a surface integral method. The near-field simulation results such as the jet centerline velocity decay and turbulence intensities as well as far-field sound were qualitatively in agreement with experimental results. The far-field sound analysis suggested significant low-frequency noise reduction for the lobed mixers, as well as significant reduction in overall sound pressure level (OASPL) in comparison with the simple confluent nozzle configurations.

9:20

3aNSb2. Airfoil flow and noise computation using monotonically integrated large eddy simulation and acoustic analogy: Effect of the grid resolution. Vasily A. Semiletov (Queen Mary, Univ. of London, Mile End Rd., London E1 4NS, United Kingdom, v.semiletov@qmul.ac.uk) and Sergey A. Karabasov (Queen Mary, Univ. of London, Cambridge, United Kingdom)

A new scalable Monotonically Integrated Large Eddy Simulation (MILES) method based on the Compact Accurately Boundary-Adjusting high-REsolution Technique (CABARET) has been applied for the simulation of unsteady flow around NACA0012 airfoil at $Re = 400{,}000$ and $M = 0.058$. The flow solution is coupled with the Ffowcs Williams–Hawkings formulation for far-field noise prediction. The computational modeling results are presented for several computational grid resolutions: 8, 16, and 32 million grid cells and compared with the experimental data available.

9:40

3aNSb3. Detached Eddy Simulation modeling and far-field trailing-edge noise estimation of a sharp-edged symmetric strut. Patrick G. Marshallsay, Laura A. Brooks, Alex Cederholm (Deep Blue Tech, Mersey Rd., Osborne, SA 5017, Australia, alex.cederholm@deepbluetech.com.au), Con J. Doolan, Danielle J. Moreau, and Cristobal Albarracin (School of Mech. Eng., The Univ. of Adelaide, Adelaide, SA, Australia)

This paper presents results of a Computational Fluid Dynamic (CFD) study of a sharp-edged symmetric flat strut at Reynolds number 500,000 based on chord at zero degrees angle of attack, and the subsequent estimation of far-field noise generated at the trailing-edge. Flow field results obtained using Detached Eddy Simulation (DES) modeling and Reynolds-averaged Navier Stokes (RANS) modeling techniques are compared with empirical wind-tunnel data. The flow is observed to be physically complex in nature, exhibiting numerical solutions that are sensitive to the mesh grid and freestream turbulence intensity. Although originally developed for use specifically with RANS-generated flow data, the RANS-based Statistical Noise Model (RSNM) technique, which estimates far-field noise from mean turbulence data via an acoustic Green's function and a statistical turbulence correlation model, is used here to estimate far-field noise spectra from both RANS and DES flow data. Far-field noise is also estimated from the DES model using the permeable surface form of the Ffowcs Williams and Hawkings (FWH) solver. The FWH estimate gives the closest match to experimental data, while the RSNM-generated noise estimate from the DES data appears to be more successful at capturing the large turbulent structures within the flow than the RANS data.

10:00

3aNSb4. Comparison of supersonic full-scale and laboratory-scale jet data and the similarity spectra for turbulent mixing noise. Tracianne B. Neilsen, Kent L. Gee, Alan T. Wall (Dept. of Phys. and Astronomy, Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, TN), and Anthony A. Atchley (Grad. Program in Acoust., Penn State Univ. Univ., State College, PA)

The broadband, partially correlated noise radiated from supersonic jets has characteristics that scale with nozzle size and flow properties. In particular, the spectral content of jet noise and the variation with angle in many cases agree with empirically derived similarity spectra for large and fine-scale components of turbulent mixing noise [Tam *et al.*, AIAA Paper 96-1716]. In previous studies, measurements made near the F-22 Raptor agreed remarkably well with the similarity spectra, with two exceptions. First, the

high-frequency slopes seen in the data were shallower than the similarity spectra at many angles. Second, the data exhibit a double frequency peak, which is absent from the similarity spectra [Neilsen *et al*., J. Acoust. Soc. Am. **132**, 1993 (2012)]. These observations are explored further by examining the spectral characteristics of noise from a different military jet and a laboratory-scale, unheated jet. In both cases, there is evidence that for supersonic cases the measured spectra are shallower than the similarity spectra due to nonlinear propagation effects. In addition, the military data support the observation that the double spectral peak is a feature of full-scale jet noise. Recommendations are made for applying the similarity spectra to predict spectral levels for full-scale jets. [Work supported by ONR.]

10:20–10:40 Break

10:40

3aNSb5. Autocorrelation analysis of military jet aircraft noise. Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astron., Brigham Young Univ., 562 N 200 E # 17, Provo, UT 84606, blaineharker@byu.net), Sally A. McInerny (Dept. of Mech. Eng., Univ. of Louisiana, Lafayette, LA), and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC)

Jet noise research has seen increased use of autocorrelation analyses to glean physical insight about the source and its radiation properties. Length scales and other features have been identified in support of models incorporating large-scale (LSS) and fine-scale (FSS) turbulent structures. In this paper, the meaningful use of autocorrelation in jet noise analysis is further examined. A key finding is that the effect of the peak frequency on autocorrelation width needs to be removed prior to making conclusions about the relative LSS and FSS contributions. In addition, the Hilbert transform is applied to create an envelope of the autocorrelation function to more consistently define a characteristic time scale. These methods are first applied to the analytical LSS and FSS similarity spectra, previously developed by Tam *et al*. [AIAA 96-1716, 1996]. It is found that the envelope of the FSS similarity autocorrelation function is more similar to that of a delta function than the LSS envelope. These curves are used to more effectively quantify FSS and LSS features in noise spectra from the F-22A Raptor. [Work supported by ONR.]

11:00

3aNSb6. A multi-objective evolutionary optimization approach to procedural flight-noise mitigation. Andrew Christian and Victor Sparrow (Grad. Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, azc144@psu.edu)

Exposure to noise is a significant problem for communities that exist near airports. The distribution of noise exposure can be positively affected by changes in the procedures that aircraft follow in the vicinity of an airport (e.g., rate of ascent, ground track, etc.). When considering such changes, a decision maker often has to weigh the objective of lower noise impact against "more practical" considerations such as fuel consumption and time-of-flight. This study presents a method of numerical optimization which seeks to find the optimal-tradeoff set (Pareto front) of flight procedures given information about an airport and the surrounding population and geography. This front will only include procedures such that an aggregate noise metric cannot be improved without detriment to a more practical objective. A contemporary multi-objective evolutionary algorithm is used as the basis of the optimization effort. Results from a simulated military airfield near Asheville, NC are shown. Ways in which decision makers are empowered by having access to a Pareto front are discussed.

11:20

3aNSb7. Preliminary analysis of acoustic intensity in a military jet noise field. Trevor A. Stout, Kent L. Gee, Tracianne B. Neilsen, Alan T. Wall, David W. Krueger (Phys., Brigham Young Univ., 688 north 500 East, Provo, UT 84606, titorep@gmail.com), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Provo, Utah)

Acoustic intensity measurements of the F-22A Raptor are analyzed as part of ongoing efforts to characterize the noise radiation from military jet aircraft. Data were recorded from a rig of microphones and an attached tetrahedral intensity probe at various locations to the sideline and aft of the aircraft. Numerical analysis of the intensity at one-third octave band center frequencies along various measurement planes and at a 23 m radius arc reveals the magnitude and directionality of the vector acoustic intensity. Differences in the trends for low-frequency and high-frequency data are discussed and, via a simple ray tracing back toward the source, interpreted in terms of source location and extent. [Work supported by ONR.]

Session 3aPA

Physical Acoustics: Borehole Acoustics Logging for Hydrocarbon Reservoir Characterization I

Said Assous, Cochair
Geoscience, Weatherford, East Leake, Loughborough LE126JX, United Kingdom

Weichang Li, Cochair
ExxonMobil Res. & Eng., 1545 Rte. 22 East, Annandale, NJ 08801

Chair's Introduction—8:55

Invited Papers

9:00

3aPA1. Complexity penalized hydraulic fracture localization and moment tensor estimation under limited model information. Gregory Ely (ECE, Tufts Univ., 8 Richard Ave., Cambridge, MA 02140, gregory.ely@tufts.edu) and Shuchin Aeron (ECE, Tufts Univ., Medford, MA)

In this paper, we present a novel technique for micro-seismic localization using a group sparse penalization that is robust to the focal mechanism of the source and requires only a velocity model of the stratigraphy rather than a full Green's function model of the earth's response. In this technique, we construct a set of perfect delta detector responses, one for each detector in the array, to a seismic event at a given location and impose a group sparsity across the array. This scheme is independent of the moment tensor and exploits the time compactness of the incident seismic signal. Furthermore, we present a method for improving the inversion of the moment tensor and Green's function when the geometry of seismic array is limited. In particular, we demonstrate that both Tikhonov regularization and truncated SVD can improve the recovery of the moment tensor and be robust to noise. We evaluate our algorithm on synthetic data and present error bounds for both estimation of the moment tensor as well as localization. Furthermore we discuss the estimated moment tensor accuracy as a function of both array geometry and fault orientation.

9:20

3aPA2. Comparison of dispersive relation of the oil well with that of the fluid cylinder and the hollow in solid. Hailan Zhang, Hanyin Cui, Weijun Lin, and Xiuming Wang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., 21 Beisihuanxilu, Beijing 100190, China, zhanghl@mail.ioa.ac.cn)

Acoustical well logging is a widely used technique in oil fields to investigate the formation outside wells and the quality of wells. The knowledge of the acoustical behavior of the wells is important to the technique and has been widely studied. The wells are usually modeled as a fluid filled cylindrical borehole in an infinite solid medium. This structure with an infinite section, sometimes called as open waveguide, is more difficult to study than the typical acoustical waveguide with finite section and free or rigid boundaries. In this presentation, the well is studied as a coupled vibration system consisted of the fluid cylinder inside the well and the solid formation outside the well. The dispersive relations of three systems, i.e., the circular fluid cylinder with rigid boundary, the circular hollow in solid medium, and the fluid filled borehole in solid, are numerically calculated and compared. The properties of the dispersive relations, such as the cutoff frequencies, normal and abnormal dispersion, are compared and discussed. [Work supported by National Natural Sciences of China, Grant No. 11134011.]

9:40

3aPA3. Space-time methods for robust slowness estimation for monopole logging while drilling. Shuchin Aeron (Dept. of ECE, Tufts Univ., 161 College Ave., Medford, MA 02155, shuchin@ece.tufts.edu), Sandip Bose (Math and Modeling, Schlumberger Doll Res., Cambridge, MA), and Henri-Pierre Valero (Acoust.-Sonic, Schlumberger K.K., Kanagawa-Ken, Japan)

In this paper, we present methods for interference cancelation for robust slowness estimation from noisy monopole logging while drilling (LWD) data. The main contributions are two fold. First, we show via tests on real data sets presence of systematic propagative interferences in monopole LWD data, which is the primary reason for loss of compression and shear semblance in the slowness time coherence (STC) processing of the LWD data. This interference in turn is mostly dominated by Stoneley type propagative component, which, unlike the main Stoneley mode, is time persistent over the entire acquisition interval. In addition, we also show that in fast formations the shear wave can significantly interfere with the compressional wave making the compressional slowness estimates quite bad. Second, based on these observations we propose a Successive Interference Cancellation (SIC) algorithm to estimate and cancel these interferences leading to STC enhancement and improved slowness estimation of the head waves. The algorithm exploits a novel representation of borehole acoustic signals using a dictionary of space-time propagators and is more robust compared to traditional slowness filtering methods, especially for the low aperture borehole acoustic array. We show the superior performance of the proposed algorithm on synthetic and real data sets.

10:00

3aPA4. Evaluation on fracturing effects in a low-permeability reservoir using acoustic logging data. Huang m. Baohua, Chen Hao, Han Jianqiang, and He Xiao (Ultrosonic Phys. and Exploration Lab, Inst. of Acoust., Chinese Acad. of Sci., No. 21, 4th Norwestern Ring RD, Haidian District, Beijing, Beijing 100190, China, huangbh@mail.ioa.ac.cn)

Low-permeability reservoirs are frequently discovered in worldwide petroleum exploration. More than 50% of oil and gas reservoirs are of low permeability. Formation fracturing technique is the most common way to develop oil production in this type of reservoirs. The fracturing effect, however, is hard to be evaluated in practice. And thus arguments always exist between constructors and geologists. We developed a favorable method to evaluate the effect from the reservoir anisotropy analysis results provided by cross dipole logging technique. The data will be measured in an open hole or a borehole when formation is before and after fractured, respectively. The formation anisotropy can be estimated from the logging data. The fracturing effects can thus be evaluated by comparing the results of perforation intervals. Small differences of anisotropy estimation results indicate failure fracturing; while good fracturing effect can be confirmed if the anisotropy of a fracturing reservoir is stronger than before. Fracturing intervals can also be predicted by the anisotropy curves of a fracturing reservoir as well as the new oil production. This approach has been applied for the evaluation of deep tight reservoirs in Daqing Oilfield and low-permeability reservoirs in Hailar. Efficient evaluation results have been obtained, which provided useful information to geologists for further explorations.

Contributed Papers

10:20

3aPA5. Wave propagation in a fluid-filled shell excited by a dipole source. Xiumei Zhang, Xiuming Wang, Hailan Zhang, and Dehua Chen (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 Northwest Ring Rd., Haidian District, Beijing 100190, China, zhangxiumei@mail.ioa.ac.cn)

Wave propagation in a fluid filled cylindrical shell excited by a dipole source is investigated for the design of a new kind of calibration pit for dipole acoustic logging tools. Based on classical elasticity wave equations, phase and group velocity dispersion curves of each mode, excitation spectra and mode contributions to wave field in the shell are presented. The dispersion curves show that the lowest mode exists in the entire frequency range with the phase velocity smaller than shear wave velocity of the shell, higher-order modes have cutoff frequencies, below the cutoff frequencies these modes are non-propagating. Analysis on mode excitation spectra and their contributions suggest that the lowest mode has potential to be used in calibrating the velocity measurement accuracy of dipole acoustic logging tools quantitatively, provided that the contributions of this mode to the wave field dominating the first arrivals gathered by a specific dipole acoustic logging tool. [Work supported by National Natural Sciences of China, Grant No. 11134011.]

10:40

3aPA6. The simulation of the seismoelectric logging while drilling based on elastic model. Xiaobo Zheng (Dept. of Astronautics and Mech., Harbin Inst. of Technol., P.O. Box 344 92# West Dazhi St., Harbin 0086150001, China, zxb3710@163.com), Xien Liu (China Oilfield Services Ltd., Sanhe, China), Hengshan Hu, and Wei Guan (Dept. of Astronautics and Mech., Harbin Inst. of Technol., Harbin, China)

In recent years, the acoustic LWD technology which is drilling and logging at the same time has been developing rapidly. However, the collar wave could cover or interfere with signals from the formation to affect the extraction of P and S wave velocities in the sonic logging. In order to solve this problem, this paper makes a research about the seismoelectric LWD response. In this study, we use the decoupling algorithm to calculate seismoelectric field in the borehole. We can obtain the elastic sound field by solving the wave equation first, and then calculate the electromagnetic field excited by the sound field by using the Pride control equations. Although using the elastic model in the calculation, we simulate the attenuation effect of pore formation by introducing quality factor and gain the pressure of the fluid within the pore by introducing the Skempton factor. Finally, this paper shows the full waveforms of the electromagnetic and acoustic fields excited by multipole source. We find that the collar wave in the electric field is significantly weakened compared with that in the acoustic pressure, in terms of its amplitude relative to the other wave groups in the full waveforms.

11:00

3aPA7. Simulation study on seismic monitoring of aquifers. Timo Lähivaara (Appl. Phys., Univ. of Eastern Finland, Kuopio, Finland), Jari P. Kaipio (Mathematics, Univ. of Auckland, Auckland, New Zealand), Nicholas F. Dudley Ward (Otago Computational Modelling Group Ltd., Kurow, New Zealand), and Tomi Huttunen (Appl. Phys., Univ. of Eastern Finland, P.O. Box 1188, Kuopio, Finland, tomi.huttunen@uef.fi)

This study focuses on developing computational tools to estimate groundwater volume from seismic measurements. The poroelastic signature from an aquifer is simulated and methods to use this signature to estimate porous properties of the permeable rock and the level of the water table are investigated. In this work, the spectral-element method (SEM) is used for solving the forward model that characterizes propagation of seismic waves. The SEM combines the accuracy of the global pseudospectral method with the flexibility of the classical finite element method. The SPECFEM-2D software is used for calculating seismic wave fields (forward + adjoint) in elastic and poroelastic media. The inverse problem is solved in the Bayesian framework, which makes efficient use of *a priori* information related to modeling and measurement uncertainties of the problem. It this study, preliminary results in the two-dimensional case with simulated data are presented.

11:20

3aPA8. Broad-band acoustic low frequency collimated beam for ultrasonic imaging. Cristian Pantea and Dipen N. Sinha (Mater. Phys. and Applications, MPA-11, Los Alamos National Lab., MS D429, Los Alamos, NM 87545, pantea@lanl.gov)

Ultrasonic and sonic imaging from a borehole is a widely used technique for hydrocarbon reservoir characterization. A typical acoustic transducer in a borehole can produce a narrow beam only at ultrasonic frequency hundreds of kHz or higher. On the other hand, high acoustic frequencies are rapidly attenuated in the formation and result in shallow penetration. To allow deeper penetration, low frequency operation is needed. We report on the development of a lower frequency broad band collimated and steerable acoustic beam source, which is sufficiently compact to fit inside a borehole and is capable of probing for rock information in the near-borehole environment. Some of the main advantages of the source presented in this study are related to: (1) the beam collimation for better spatial resolution, (2) beam steerability (360 degree in azimuth and inclination), (3) the broad-bandwidth (20–120 kHz) and (4) specific pulse shape for simpler signal processing and data analysis. Laboratory experimental data simulating open and cased wellbores will be presented.

11:40–12:00 Panel Discussion

Session 3aPP

Psychological and Physiological Acoustics: Auditory Physiology and Modeling (Poster Session)

Magdalena Wojtczak, Chair
Psychology, Univ. of Minnesota, 1237 Imperial Ln, New Brighton, MN 55112

Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

3aPP1. The medial olivary complex reflex strength of children with auditory processing disorders. Sangeeta Kamdar and Su-Hyun Jin (Commun. Sci. and Disord., Univ. of Texas at Austin, 1 University Station, A1100, Austin, TX 78712, kamdarsc@gmail.com)

The present investigation is designed to examine speech understanding in noise and the strength of medial olivary complex (MOC) reflex in children diagnosed with auditory processing disorder (APD). APD is a dysfunction associated with limited auditory processing of sounds. Individuals with APD do not show any peripheral hearing impairment but have difficulty understanding speech especially in the presence of noise. Recent neurophysiological studies suggest that the efferent system also make an important contribution. One of the most peripheral parts of the auditory efferent system is MOC system which projects from the auditory brainstem to the cochlea. The MOC system has been known to play an important role protecting the auditory system from intense noise and affecting tone detection or speech perception in noise [Micheyl and Collet (1996), Kumar and Vanaja (2004)]. The strength of this efferent feedback system can be assessed non-invasively through the contralateral suppression of distortion product otoacoustic emissions. It is hypothesized that children with APD show weaker MOC reflex strength compared to the normal peers. Based on the test results, possible efficient testing protocol and intervention program for this special population will be discussed.

3aPP2. Evaluating the role of efferent inhibition on cochlear responses: Simultaneous psychophysical and otoacoustic emission measurements. Simon Henin and Glenis Long (Ph.D. Program in Speech-Language-Hearing Sci., The Grad. Ctr., The City Univ. of New York, 365 Fifth Ave., New York, NY 10016, shenin@gc.cuny.edu)

The auditory system continuously adapts to changes in the acoustic environment. Behavioral experiments in humans have demonstrated that changes in the acoustic environment produce dynamic changes in perception, for example, increases in thresholds in the presence of background noise. This dynamic change in the auditory system is hypothesized to be mediated by efferent feedback from the olivocochlear system. The effect of efferent inhibition on cochlear mechanics was investigated using a simultaneous psychoacoustics and otoacoustic emissions (OAEs) task using identical stimulus conditions. Cochlear responses to short tone-burst stimuli were analyzed under various masking conditions. Robust modification of cochlear responses to the short tone-burst stimuli was observed during contralateral acoustic stimulation and during long-duration ipsilateral masking. Concomitant changes in perceptual thresholds and OAEs are consistent with the hypothesis that both stem from efferent activation. This novel paradigm provides simultaneous perceptual and physiological estimates of cochlear-based efferent activation in the same human subjects.

3aPP3. Modeling psychophysical gain reduction effects as a function of precursor duration. Elin Roverud and Elizabeth A. Strickland (Speech, Lang., and Hearing Sci., Purdue Univ., 2501 Soldiers Home Rd., apt 16G, West Lafayette, IN 47906, eroverud@purdue.edu)

It is known that a forward masker can make threshold for a signal poorer, but the mechanisms underlying this psychophysical effect are not well-understood. One theory, the temporal window model (TWM), proposes that masker and signal excitation are integrated within a temporal window. An additional mechanism may be cochlear gain reduction by the medial olivocochlear reflex, a sluggish sound-evoked reflex. In our laboratory, we have shown evidence of gain reduction in forward masking results. We measure off-frequency growth of masking to estimate the cochlear input/output (I/O) function. A precursor is introduced which reduces the gain of the I/O function. In the present study, we examine this gain reduction effect as a function of precursor duration for on- and off-frequency precursors. From a gain reduction perspective, a long on-frequency precursor may reduce gain for itself, while the off-frequency precursor, assumed to be linear at the signal frequency, would not. The TWM, however, does not predict differences in trends with duration for the two precursor frequencies. In our modeling, we have incorporated a gain reduction module into the TWM. We will compare predictions of the results with the standard TWM and the TWM with gain reduction. [Research supported by NIH(NIDCD)R01 DC008327.]

3aPP4. The effect of the medial olivocochlear reflex on click-evoked otoacoustic emissions during psychoacoustic forward-masking tasks. Jordan A. Beim, Magdalena Wojtczak, and Andrew J. Oxenham (Psychology, Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455, beimx004@umn.edu)

Measurements of otoacoustic emissions in animals have shown that the effects of efferent activation are greater in attentive than in anesthetized animals suggesting that the medial olivocochlear reflex (MOCR) effects can be modulated by attention. In this study, the effect of efferent activation was measured in humans using click-evoked otoacoustic emissions while listeners were performing a psychoacoustic forward-masking task. Each trial within a block started with a sequence of 50-dB pSPL clicks presented at a rate of 40 Hz that were followed by a 200-ms harmonic-complex masker. The masker was immediately followed by a 10-ms tonal probe and another click train. The listeners' task was to detect the probe. A constant stimuli method was used to measure performance in the forward-masking task, with the probe presented at seven randomized levels around the predetermined masked threshold. Catch trials were dispersed randomly throughout the block. Click trains before and after the masker-signal segments were recorded from the ear canal and analyzed to extract effects of efferent activation at different levels of difficulty of the psychoacoustic task. The results will be discussed with respect to the role of attention and the role of the MOCR in forward masking. [Work supported by NIH grant R01DC010374.]

3aPP5. Human otoacoustic emissions generated by active outer hair cells. Reinhart Frosch (ETH and PSI (retired), Sommerhaldenstrasse 5B, Brugg 5200, Switzerland, reinifrosch@bluewin.ch)

In the present study, human otoacoustic emissions (OAEs) documented in the literature are shown to agree with predictions based on the hypothesis that the main OAE sources are active cochlear outer hair cells (OHCs) of three different functional categories, namely (1) OHCs enabled by oscillating internal organ-of-Corti resonators (IOCRs) to feed mechanical energy into forward-traveling cochlear waves generated by the acoustic stimuli used, (2) OHCs driving spontaneous localized feedback-generated cochlear-

partition vibrations involving standing evanescent sound-pressure waves in the liquids above and below the partition, and (3) OHCs causing nonlinear restoring forces enabling pairs of stationary tones to generate distortion products (DPs). The corresponding predictions of OAE properties are based on cochlear maps, i.e., on certain functions x(f), where f is the frequency of a tone and x is a related distance from the cochlear base, measured along the cochlear channel.

3aPP6. Total and component distortion product otoacoustic emission analysis in persons with induced negative middle ear pressure. Suzanne Thompson, Glenis Long, and Simon Henin (Ph.D. Program in Speech-Language-Hearing Sci., The Grad. Ctr. of the City Univ. of New York, 40 Hathaway Dr., Garden City, NY 11530, sthompson1@gc.cuny.edu)

Distortion product otoacoustic emissions (DPOAEs) are generated when two primary tones (f_1, f_2 with $f_2 > f_1$) are presented simultaneously to the ear. Inter-modulation between primary tones produces distortion products at predictable frequencies not present in the original signal (e.g., $2f_1 - f_2$). In persons with negative middle ear pressure (NMEP), the tympanic membrane is retracted and pulled inward, compressing structures in the middle ear. NMEP is expected to modify DPOAE level and phase. Performing the Toynbee maneuver can artificially induce NMEP. Changes in DPOAE primary f_1 and f_2 level and phase and energy reflectance measures were used to confirm that 8 subjects had artificially induced NMEP using the Toynbee maneuver. DPOAEs were obtained using 1 s/octave duration logarithmic frequency sweeping primaries, $f_2/f_1 = 1.22$ producing $2f_1 - f_2$ from 320–2560 Hz, L1 = L2 = 65, 70, 75 dB SPL. There was a significant effect of condition, primary level, and frequency on total DPOAE and component amplitude. Separation of DPOAE components allowed improved detection of NMEP effects on DPOAE amplitude.

3aPP7. Effect of percussion impulse sounds on hearing. David Pazen, Dirk Beutner, and Martin Walger (Dept. of Otorhinolaryngol., Head and Neck Surgery, Univ. Hospital of Cologne, Kerpener Str. 62, Cologne 50937, Germany, david.pazen@uk-koeln.de)

Impulse-like sounds with high peak pressure levels emitted by percussion instruments often occur in music. The potentially harmful effect to the inner ear caused by these sounds is not fully understood. Solely the physical description of impulses like peak pressure level, etc. is not considered meaningful enough to assess their hazard. Therefore, the Auditory Hazard Assessment Algorithm for Humans (AHAAH), which is based on a physiological model, seems to be more appropriate. Another quantity to assess auditory hazard is the change of otoacoustic emissions (OAEs) after a sound exposure. It allows the individual detection of small physiological changes of the outer hair cells before any serious damages and threshold shifts may occur. In a pilot experiment the sounds of several percussion instruments have been measured at the players' ears in a regular rehearsal situation. To assess the auditory hazard of the measured sounds the players' OAEs have been measured before and after playing on the instruments. Despite peak levels of about 140 dB, the OAEs did not change significantly and the AHAAH rated nearly all sounds as harmless. It can be concluded that impulse sounds in music with high peak levels are not necessarily hazardous.

3aPP8. Efficient estimates of cochlear hearing loss parameters in individual listeners. Michal Fereczkowski (Elektro, DTU, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, mfer@elektro.dtu.dk), Morten L. Jepsen (R&D, Widex A/S, Lynge, Denmark), and Torsten Dau (Elektro, DTU, Kgs Lyngby, Denmark)

It has been suggested that the level corresponding to the knee-point of the basilar membrane (BM) input/output (I/O) function can be used to estimate the amount of inner- and outer hair-cell loss in listeners with a moderate cochlear hearing impairment [e.g., Plack *et al*., (2004)]. In the present study, results from forward masking experiments based on temporal masking curves [TMC; Nelson *et al*. (2001)] are presented and used to estimate the knee-point level and the compression ratio of the I/O function. A time-efficient paradigm based on the single-interval-up-down method [SIUD; Lecluyse and Meddis (2009)] was used instead of an alternative forced-choice paradigm. In contrast with previous studies, the present study used only on-frequency TMCs to derive estimates of the knee-point level. Further, it is explored whether it is possible to estimate the compression ratio using only on-frequency TMCs. Ten normal-hearing and 10 hearing-impaired listeners (with mild-to-moderate sensorineural hearing loss) were tested at 1, 2, and 4 kHz. The results showed a reasonable reliability and an increased time-efficiency compared to AFC-based results and may be suitable for individualized hearing-aid fitting.

3aPP9. Relationship of distortion product otoacoustic emission components to psychoacoustic measures of noise induced hearing loss. Gavin Coad (Section of Audiol., The Univ. of Auckland, Morrin Rd., Glen Innes, Auckland 2012, New Zealand, g.coad@auckland.ac.nz), Glenis R. Long (Speech-Lang.-Hearing Program, CUNY, New York, NY), David Welch, and Peter R. Thorne (Section of Audiol., The Univ. of Auckland, Auckland, New Zealand)

Distortion product otoacoustic emissions (DPOAEs) have promise as a tool to detect and monitor cochlear hair cell damage in noise induced hearing loss (NIHL). However, variability in OAE amplitudes and thresholds across individuals limits their potential for detecting the early impact of NIHL. The DPOAE has two components, one generated at the point of interaction between the primaries (generator), the second reflected from the characteristic place of the DPOAE frequency (reflected). There has been increasing interest in looking at the components of the DPOAE and how they correlate with injury. A total of 109 men, including 57 with varying degrees of occupational noise exposure, were tested. We recorded DPOAEs using logarithmically swept pure tones and extracted the components of the DPOAE using a least-squares fit approach. Pure-tone audiometry was conducted at 1.5 and 4 kHz with 1dB resolution. Compressive nonlinearity was evaluated psychoacoustically with 1.5 and 4 kHz pure-tones using Schroeder phase maskers. Overall there are associations between the generator component and the psychoacoustic measures: stronger DPOAE amplitudes were associated with better audiometric thresholds and greater difference in Schroeder masking threshold. Associations were stronger at 4 than 1.5 kHz. Similar, but weaker effects were observed for the reflected component. Findings suggest that separation of DPOAE components enhances the assessment of cochlear noise-induced injury.

3aPP10. Correlations between noninvasive and direct physiological metrics of auditory function in chinchillas with noise-induced hearing loss. Kenneth S. Henry (Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907, kshenry@purdue.edu), Sandra F. Snyder (Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN), and Michael G. Heinz (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

Noninvasive physiological tools for assessing auditory function in humans can provide valuable information when behavioral tests are not possible. Furthermore, these tools hold promise to provide greater insight into underlying cochlear pathologies. In this study, we used noninvasive distortion product otoacoustic emissions (DPOAEs) and auditory brainstem responses (ABRs) to estimate changes in auditory function in chinchillas with noise-induced hearing loss. Two aspects of cochlear function, sensitivity (threshold) and frequency selectivity (tuning), were measured directly using neurophysiological recordings from auditory-nerve (AN) fibers to assess the predictive value of DPOAEs and ABRs. Both DPOAE amplitude and ABR threshold were well correlated (R-square ~0.5) with AN fiber threshold near stimulus frequency. For DPOAEs, the correlation was strongest for cochlear function near F2. Correlations of both noninvasive metrics with AN tuning were weaker but statistically significant. The relatively weak correlation between DPOAE amplitude and AN tuning was unexpected because both measures are tied to the integrity of outer hair cells. Alternatively, previous results suggest ABR latency may be more predictive of AN tuning. Ultimately, DPOAEs may be a more practical clinical and research screening tool for hearing loss than ABRs due to shorter recording time. [Research supported by NIH (NIDCD) F32-DC012236 and R01-DC009838.]

3aPP11. Effects of inner hair cell damage on temporal coding. David R. Axe and Michael G. Heinz (Biomedical Eng., Purdue Univ., 320 Brown St., Apt. 713, West Lafayette, IN 47906, davidraxe@gmail.com)

It is widely believed that the neural patterns of temporal coding within the auditory periphery and CNS change following cochlear hearing loss, and a number of recent studies have aimed to more fully understand and characterize these changes. In these studies, noise exposure has been a common method for inducing hearing loss in animal models. Unfortunately, its effects are nonspecific, affecting both inner and outer hair cells as well as the surrounding tissues. Because of this mixed hair-cell damage, it has been difficult to tease apart the specific effects that each of these pathologies has on temporal coding. In the present study, we used the chemotherapy drug carboplatin to induce inner hair cell (IHC) specific lesions in the cochleae of chinchillas. Using acoustically evoked potentials and acute single fiber recordings from the auditory nerve, in parallel with computational modeling, we have investigated the effects of IHC damage on temporal coding. Preliminary findings in carboplatin exposed chinchillas, which show near-normal ABR thresholds, showed a decrease in ABR amplitudes at high sound levels, as well as a decrease in the strength of both envelope and fine-structure coding in frequency following responses. [Research supported by NIH (NIDCD) R01-DC009838 and T32-DC00030.]

3aPP12. Prolonged low-grade noise exposure induces aging-like functional and structural changes in cortical auditory pathways. Brishna Kamal, Lydia Ouellet, and Etienne de Villers-Sidani (Neurology, McGill Univ., 3801 University St., Rm. 736, Montreal, QC, Canada, brishnak@gmail.com)

Age-related impairments in the primary auditory cortex (A1) include poor tuning selectivity, neural desynchronization, and degraded responses to low-probability or "oddball" sounds. These changes have been for the most part attributed to reduced inhibition in the aged brain. Since many of these changes can be partially reversed with auditory training, it has been speculated that they might not be purely degenerative but might rather represent negative plastic adjustments to noisy or distorted auditory inputs. To test this hypothesis, we exposed young adult rats to low-grade broadband noise for 6 weeks and then compared the effect of this exposure on several aspects of A1 function and structure. We found that the impact of noise exposure on A1 tuning selectivity and responses to oddball tones was almost undistinguishable from the effect of natural aging. These changes were paralleled by alterations in A1 inhibitory interneuron populations in the exposed group. Moreover we found that noise exposure reduced the anatomical and functional connectivity of A1 to downstream cortical fields. Most of these changes reversed after returning to a non-noisy environment. These results support the hypothesis that age-related changes in cortical auditory pathways might have a strong activity-dependent component, making them potentially preventable and reversible.

3aPP13. How broadband speech may avoid neural firing rate saturation at high intensities and maintain intelligibility. Richard Warren, James Bashford, and Peter Lenz (Psychology, Univ. of Wisconsin-Milwaukee, P.O. Box 413, Milwaukee, WI 53201, rmwarren@uwm.edu)

While broadband speech may remain perfectly intelligible at levels exceeding 90 dB, narrowband speech intelligibility (e.g., 2/3-octave passband centered at 1.5 kHz) may decline by 25% or more at moderate intensities (e.g., 75 dB). This "rollover" effect is substantially reduced, however, when a speech band is accompanied by flanking bands of white noise [Bashford *et al*., J. Acoust. Soc. Am. **117**, 365–369 (2005)], suggesting that lateral suppression helps preserve broadband speech intelligibility at high levels. The present study found that when noise flankers were presented individually at a low spectrum level (-30 dB relative to the speech) only the higher-frequency flanker produced a significant intelligibility increase. However, the lower-frequency flanking noise did produce an equivalent increase when its spectrum level was raised 10 dB. This asymmetrical intensity requirement for noise flankers links the effective dynamic range of speech intelligibility to reported characteristics of both lateral (two-tone) suppression of auditory nerve (AN) fiber activity and lateral inhibition of secondary cells of the cochlear nucleus. These and other observations will be discussed in the broader context of how various auditory mechanisms help preserve speech intelligibility at high intensities by reducing firing rate saturation. [Work supported by NIH.]

3aPP14. The computational prediction of masking thresholds for ecologically valid interference scenarios. Khan Baykaner, Christopher Hummersone, Russell Mason (Inst. of Sound Recording, Univ. of Surrey, 26 Shepherds Hill, Guildford GU2 9RY, United Kingdom, ee51kb@surrey.ac.uk), and Søren Bech (Bang & Olufsen, Struer, Denmark)

Auditory interference scenarios, where a listener wishes to attend to some target audio while being presented with interfering audio, are prevalent in daily life. The goal of developing an accurate computational model which can predict masking thresholds for such scenarios is still incomplete. While some sophisticated, physiologically inspired, masking prediction models exist, they are rarely tested with ecologically valid programs (such as music and speech). In order to test the accuracy of model predictions human listener data is required. To that end a masking threshold experiment was conducted for a variety of target and interferer programs. The results were analyzed alongside predictions made by the computational auditory signal processing and prediction model described by Jepsen *et al*. (2008). Masking thresholds were predicted to within 3 dB root mean squared error with the greatest prediction inaccuracies occurring in the presence of speech. These results are comparable to those of the model by Glasberg and Moore (2005) for predicting the audibility of time-varying sounds in the presence of background sounds, which otherwise represent the most accurate predictions of this type in the literature.

3aPP15. A computational model of spatial tuning in the auditory cortex in response to competing sound sources. Junzi Dong, Steven Colburn, and Kamal Sen (Boston Univ., 18 Medfield St., Apt. 1, Boston, MA 02215, junzid@bu.edu)

Single neurons in the auditory midbrain are sharply tuned to preferred directions, while cortical neurons show broader tuning [King *et al*., Hearing Research (2007)] Recent experiments on cortical responses in birds revealed the emergence of spatial tuning to multiple competing sounds in the cortex [Maddox *et al*., PLoS Biol. (2012)]. In this situation, cortical neurons show broad tuning to single-location stimuli, but develop sharper spatial preference in response to a second competing noise source. We have developed a computational model to match these physiological data. The model takes binaural inputs containing sounds from two locations, and outputs cortical spike trains that emphasize one pre-defined location. In the model, sharply tuned midbrain neurons synapse onto their corresponding interneurons, which then innervate cortical neurons generating the final output. In the presence of stimulus from a pre-defined preferred location, corresponding interneurons actively suppress sources from a pre-defined non-preferred location. The model achieves spatial tuning by performing robustly when the target and masker locations match the pre-defined preferred and non-preferred directions. Looking across cortical neurons tuned to different target and masker combinations, the model provides a mechanism by which binaural inputs from a noisy environment can be separated into independent "streams" based on locations.

3aPP16. Combining the outputs of functional models of organs responsible for binaural cue decoding. Marko O. Takanen (Dept. of Signal Processing and Acoust., Aalto Univ. School of Elec. Eng., Otakaari 5A, Espoo 02015, Finland, marko.takanen@aalto.fi), Olli Santala (Dept. of Signal Processing and Acoust., Aalto Univ. School of Elec. Eng., Helsinki, Finland), and Ville Pulkki (Dept. of Signal Processing and Acoust., Aalto Univ. School of Elec. Eng., Espoo, Finland)

The binaural cue decoding in the human auditory pathway occurs in the medial superior olive (MSO) and the lateral superior olive (LSO). The MSO is sensitive to interaural time difference (ITD), whereas the LSO is sensitive to interaural level difference as well as to ITD at low frequencies. Functional models of such organs have been presented previously [Pulkki and Hirvonen, Acta Acoustica united with Acoustica **95**, 883–900 (2009)]. However, the outputs of those models are not as such applicable as the level of the output varies for a point-like broadband stimulus depending on the frequency and between the models. Here, a method is presented to combine the outputs of such models and an additional model of the MSO designed to decode directional cues based on broadband envelope time-shifts between the ear canal signals. Applicable cues are obtained by mapping the outputs into azimuth direction values following the idea of self-calibration, and by favoring the cue values suggesting more lateral directions. Furthermore, it is shown how the resulting directional cues can be applied to form a binaural activity map, and that the activity map corresponds to the human perception in several scenarios of psychoacoustical experiments.

3aPP17. Interaction between resonant scaling and fundamental frequency on codings in auditory system. Toshie Matsui (Dept. of Otorhinolaryngol. - Head and Neck Surgery, Nara Med. Univ., Shijo-cho 840, Kashihara City 634-8522, Japan, tomatsui@naramed-u.ac.jp) and Minoru Tsuzaki (Faculty of Music, Kyoto City Univ. of Arts, Kyoto city, Japan)

It has been modeled that the auditory system encodes acoustic signals into two independent informations. One is tonotopic information reflecting the frequency response characteristics of the basilar membrane, and another is periodicity information reflecting the temporal patterns of phase-locked auditory nerve firing. Based on the previous study about size perception of

sound source, a hypothesis can be proposed that the tonotopic and periodicity information are combined into an internal "two-dimensional representational plane." The current study tested that this hypothesis by conducting an experiment to recognize the transition pattern of vowel-like sounds on the plane. Listeners were able to recognize the transition patterns on the axes of tonotopic and periodicity information exclusively. However, they were unable to follow the trajectories of sounds in the two-dimensional representational plane. The vowel-like sounds were simulated by a computational model of auditory system, AIM. While the "excitation pattern" displayed transition patterns corresponding to the manipulation of resonant scaling (RS) only, "Summary SAI" displayed both of patterns of fundamental frequency (F0) and RS. It suggests the need to confirm whether RS represented in periodicity information is used for size perception.

3aPP18. A simulation of neural coding and auditory frequency analysis. Robert A. Houde (Ctr. for Commun. Res., 35 Rensselaer Dr., Rochester, NY 14618, rahoude@gmail.com), James M. Hillenbrand (Speech Pathol. and Audiol., Western Michigan Univ., Kalamazoo, MI), Robert T. Gayvert (Ctr. for Commun. Res., Rochester, NY), and John F. Houde (Dept. of Otolaryngol., Univ. of California at San Francisco, San Francisco, CA)

Our understanding of the neural mechanisms underlying the very fine auditory frequency discrimination exhibited by listeners remains far from complete. To investigate this question we developed a functional model of the cochlear process in sufficient detail to allow the simulation of the principal characteristics of the cochlea's response to multi-tone and noise stimuli over a wide range of input levels. The model simulates level-dependent changes in frequency selectivity, combination-tone distortion, tone-on-tone suppression and masking, adaptation, and critical-band masking. The model is structured as 3000 channels, each consisting of a basilar membrane bandpass filter and inner-hair cell assembly. Input to each channel is the stapes displacement signal, and the output consists of ten independent stochastic point processes that are transmitted to the CNS on auditory-nerve fibers (ANFs). Our main purpose is to address these questions: (1) What narrowband spectrum information is available in the cochlea output? (2) How is this information encoded on the ANFs? (3) How might it be decoded in the CNS? An analysis of ensemble coding of the cochlear output showed that the precision (signal-to-noise ratio) of the information decoded by the CNS frequency analysis is directly related to the bandwidth of the basilar membrane filters.

3aPP19. The effect of compression on tuning estimates in a simple nonlinear auditory filter model. Márton Marschall, Ewen MacDonald, and Torsten Dau (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, mm@elektro.dtu.dk)

Behavioral experiments using auditory masking have been used to characterize frequency selectivity, one of the basic properties of the auditory system. However, due to the nonlinear response of the basilar membrane, the interpretation of these experiments may not be straightforward. Specifically, there is evidence that human frequency-selectivity estimates depend on whether an iso-input or an iso-response measurement paradigm is used [Eustaquio-Martin *et al.* (2011)]. This study presents simulated tuning estimates using a simple compressive auditory filter model, the bandpass nonlinearity (BPNL), which consists of a compressor between two bandpass filters. The BPNL forms the basis of the dual-resonance nonlinear (DRNL) filter that has been used in a number of modeling studies. The location of the nonlinear element and its effect on the estimated tuning in the two measurement paradigms was investigated. The results show that compression leads to (i) a narrower tuning estimate in the iso-response paradigm when a compressor precedes a filter, and (ii) a wider tuning estimate in the iso-input paradigm when a compressor follows a filter. The results imply that if the DRNL presents a valid model of the basilar membrane, then compression alone may explain a large part of the behaviorally observed differences in tuning between simultaneous and forward-masking conditions.

3aPP20. Frequency-domain analysis of cochlear gain reduction due to disruptions in the outer hair cell feedback loop. Yi-Wen Liu, Kuang-Yi Lin, and Yong-Zing Chen (Elec. Eng., National Tsing Hua Univ., 101 Kuang-Fu Rd. Sec 2, Delta Bldg. Rm. 828, Hsinchu 30013, Taiwan, ywliu@ee.nthu.edu.tw)

A frequency domain equivalent model was implemented to match the small-signal responses produced by a time-domain cochlear mechanics model [Liu and Neely (2010)]. In the present model, the outer hair cell feedback is characterized by physical parameters, including a hair-bundle transduction ratio (HBTR) and a prestin-associated capacitance (PAC). When either of them is reduced, the model predicts lowered magnitude responses along the cochlea; however, HBTR and PAC seem to play different roles in facilitating cochlear amplification. It appears that the gain drops more drastically with respect to reduction in the HBTR, whereas the degradation is more graceful with respect to lack of prestin. Simulation also suggests that HBTR is more crucial for high-frequency hearing whereas PAC boosts up low frequency responses. Finally, we present attempts on modifying the frequency-domain model iteratively to simulate nonlinear responses. The saturating cochlear responses to high-intensity tones can be predicted by the frequency domain model as long as HBTRs are adjusted appropriately. Currently, high precision is achieved (as compared against time-domain simulation) by usage of Fourier series analysis in determining the HBTR adjustment factor. This numerical approach has a potential to accelerate simulation by orders of magnitude. Its physical meaning will also be discussed.

3aPP21. A physiology-based auditory model elucidating the function of the cochlear amplifier and related phenomena. Part I: Model structure and computational method. Herbert Hudde and Sebastian Becker (Inst. of Commun. Acoust., Ruhr-Univ. Bochum, Bochum D-44780, Germany, herbert.hudde@rub.de)

An auditory model "PhyBAM" is presented, which in the long term aims at reproducing the human auditory perception. In recent years, the awareness has grown that many perceptive features have their origin in the peripheral ear, above all in the cochlea. In the present stage, PhyBAM is actually just a model of the peripheral ear. To simulate the perception of arbitrary sound signals, the signal processing occurring in the cochlea has to be formulated close to the physiological basis. Even so the model must be kept as simple as possible for the given aim. As a compromise, PhyBAM is set up as an ordinary circuit model. In this first of two associated papers, the model structure and the computational methods are presented. The model covers ear canal, middle ear, and cochlea. The cochlea model is by far the most sophisticated part. To include the unsymmetrical conditions at both cochlear windows and the resulting common and differential modes, a two-canal circuit is used. The main challenge is the implementation of the cochlear amplifier on the basis of measured tuning curves and otoacoustic emissions. Finding an appropriate model structure and proper parameters turns PhyBAM into an instrument of cochlear research.

3aPP22. A physiology-based auditory model elucidating the function of the cochlear amplifier and related phenomena. Part II: Model parameters and simulations. Sebastian Becker and Herbert Hudde (Inst. of Commun. Acoust., Ruhr Univ. Bochum, Bldg. ID 2/261 Ruhr-Universität Bochum, Bochum 44780, Germany, sebastian.becker-2@rub.de)

In this second of two associated papers, the properties of the physiology-based auditory model are investigated. This includes finding of appropriate parameters and simulating various responses. In the end the model is intended to reproduce the human ear, hence human data is used for fitting. Only the trend of active tuning curves is based on chinchilla measurements, as human data is not available. To achieve such tuning curves the cochlea amplifier feeds energy into the system basal to the characteristic place, resulting in a locally restricted negative real part of the basilar membrane impedance. Realistic level dependent tuning curves show a reasonable input-output function and a maximum cochlear gain of 55 dB. The growth of distortion product otoacoustic emissions is consistent with measurements and shows a slope of 0.5 dB/dB. The physiology-based model approach shows the origin of the distortion products within the overlap region of the stimuli and elucidates the propagation within cochlea. As reflections are a dominant factor in the generation of transient evoked otoacoustic emissions, parameters need a certain degree of roughness to achieve results corresponding to measurements. In spite of its simplicity the model is able to reproduce a variety of cochlea results with one parameter set.

3aPP23. The coda of the transient response in a sensitive cochlea. Yizeng Li and Karl Grosh (Mech. Eng., Univ. of Michigan - Ann Arbor, 2350 Hayward Ave., Ann Arbor, MI 48109, yizengli@umich.edu)

In a sensitive cochlea, the basilar membrane (BM) velocity response due to transient external acoustic excitation or to localized transient internal bipolar electrical excitation gives rise not only to a primary impulse response, but also to a coda of delayed secondary responses (sometimes called echoes or

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ringing) with varying amplitudes but similar spectral content around the best frequency of the measurement location. The coda is physiologically vulnerable, disappearing when the cochlea is compromised even slightly. The multicomponent sensitive response is not yet completely understood. We use a mathematical model to describe how the response at the point of excitation can be traced back to three sources. Surprisingly, the first BM response is due to a fast wave emergent from the point of excitation, reflected by the stapes and then repropagated (in amplified fashion) as a traveling wave back to the point of excitation; the second is due to a reverse, slow, traveling wave, which is likewise reflected at the stapes back to the measurement location by the stapes. The coda is also due to systematic (not random) perturbations of the organ of Corti properties. Implications for normal hearing and for the interpretation of otoacoustic emissions are discussed.

3aPP24. Wave finite element analysis of an active cochlear model. Guangjian Ni and Stephen J. Elliott (Inst. of Sound and Vib. Res., Univ. of Southampton, SPCG, ISVR, Univ. of Southampton, Southampton, Hampshire SO17 1BJ, United Kingdom, S.J.Elliott@soton.ac.uk)

The wave finite element method has previously been used to understand the various types of wave in a passive cochlear model. In this paper, the model is extended to give an initial representation of an active cochlea by making the real and imaginary components of the Young's modulus, that defines the local dynamics of the plates representing the basilar membrane, position and frequency dependent. At a given excitation frequency, the distribution of the Young's modulus is chosen so that the mechanical impedance of the plate elements correspond to that obtained from a lumped parameter model of the active cochlea. The types of wave predicted in this representation of the active cochlea are similar to those observed for the passive cochlea model and consist of a fast wave, a slow wave and a large number of higher-order fluid modes, which are evanescent. Although the results of the full finite element analysis for this active model are very different from the passive one, the decomposition into wave components still shows that the slow wave dominates the response along most of the cochlear length, until the response peaks, when a number of higher-order fluid modes are locally excited.

3aPP25. Can cochlear mechanics contribute to amplitude modulation perception? Jungmee Lee and Sumitrajit Dhar (Roxelyn and Richard Pepper Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, jmlee6@msn.com)

Amplitude modulation (AM) detection has been successfully used as a psychophysical measure of auditory temporal processing. Our understanding of the role of the auditory periphery in processing AM signals is emerging through physiological and psychophysical studies. Unfortunately, direct physiological estimates of the cochlea's mechanical response to AM signals are not obtainable in humans. This study tries to fill this critical gap in knowledge by exploring the relationship between perception (through psychophysical AM detection) and mechanics (through otoacoustic emissions). Psychometric function for AM perception was measured for a 2-kHz carrier frequency and 10-Hz modulation frequency (f_m). Distortion product otoacoustic emissions (DPOAEs) were recorded with amplitude-modulated f_1 with $f_m = 10$ Hz and steady-state f_2. The frequencies of f_1 and f_2 were chosen to yield a $2f_1 - f_2$ DPOAE around 2 kHz near a peak in the fine structure. The ratio between the DPOAE pressure at $2f_1 - f_2$ and that of the sidebands separated by f_m (AMOAE depth) was calculated as a function of different modulation depths. Results indicate that there might be a correlation between AM perception performance and AMOAE magnitude, suggesting that cochlear mechanics might play a role for AM perception. [Work supported by the Knowles Hearing Center and Northwestern University.]

3aPP26. Effects of spontaneous otoacoustic emissions on frequency discrimination. Rói Hansen, Sébastien Santurette (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, Ørsteds Plads, DTU Bygning 352, Kgs. Lyngby 2800, Denmark, ses@elektro.dtu.dk), and Sarah Verhulst (Auditory Neurosci. Lab., Boston Univ., Boston, MA)

When an external tone is presented in proximity to the frequency of a spontaneous otoacoustic emission (SOAE), the SOAE typically synchronizes to the external tone, a phenomenon known as "entrainment". As the tone moves further away from the SOAE frequency, beating patterns between the SOAE and the pure tone occur [Long, Hear. Res. **119** (1998)]. This study investigated perceptual consequences of SOAE beating and entrainment on the difference limen for frequency (DLF), which has been found to improve near an SOAE. SOAE entrainment patterns were obtained for six subjects with a strong SOAE in the ipsilateral ear and no SOAE in the corresponding frequency range of the contralateral ear. Hearing thresholds and DLFs were measured ipsi- and contralaterally for nine frequencies covering the entrainment and beating regions of the SOAE. DLFs systematically improved in the entrainment region, worsened when beating occurred, and improved again for frequencies further away from the SOAE. No improvement in DLF was found in any of the contralateral ears tested, suggesting that the effect is of peripheral, rather than of central, origin. The results contradict an earlier hypothesis stating that DLF performance near SOAE frequencies is governed by a central oversensitivity to the SOAE frequency.

3aPP27. Temporal integration near threshold fine structure–The role of cochlear processing. Bastian Epp (Ctr. for Hearing and Speech Sci., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Rm. 118, Lyngby 2800, Denmark, bepp@elektro.dtu.dk), Manfred Mauermann (Med. Phys., Univ. of Oldenburg, Oldenburg, Germany), and Jesko L. Verhey (Dept. of Experimental Audiol., Otto-von-Guericke Univ., Magdeburg, Germany)

The hearing thresholds of normal hearing listeners often show quasi-periodic variations when measured with a high frequency resolution. This hearing threshold fine structure is related to other frequency specific variations in the perception of sound such as loudness and amplitude modulated tones at low intensities. The detection threshold of a pulsed tone also depends not only on the pulse duration, but also on the position of its frequency within threshold fine structure. The present study investigates if psychoacoustical data on detection of a pulsed tone can be explained with a nonlinear and active transmission line cochlea model. The model was successfully applied to other psychoacoustical data at low intensities, various types of otoacoustic emissions and physiological data. The simulations show differences in detection thresholds for tones placed in a minimum or a maximum of the fine structure, but lack a decrease of thresholds with increased pulse duration. The model was extended by including a temporal integrator which introduces a low-pass behavior of the data with different slopes of the predicted threshold curves, producing good agreement with the data. On the basis of the model simulations, it will be discussed to which extent temporal and spectral aspects contribute to the data.

3aPP28. Accuracy of synchrony judgment between two pulses: Effects of variations in cochlear delay amount. Eriko Aiba (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol., 1-8-31 Midorigaoka, Ikeda 563-8577, Japan, aiba.eriko@aist.go.jp), Minoru Tsuzaki (Faculty of Music, Kyoto City Univ. of Arts, Kyoto, Japan), Noriko Nagata (School of Sci. and Technolog., Kwansei Gakuin Univ., Sanda, Japan), and Seiji Nakagawa (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol., Ikeda, Japan)

The cochlear delay shifts the arrival of lower-frequency components of an auditory signal slightly but systematically behind that of higher-frequency components. Therefore, even if all of the components of a complex tone physically begin simultaneously, their temporal relation is not preserved at the cochlear level. In our previous study, the accuracy of synchrony judgment was measured using two types of chirps (compensated and enhanced chirps) and a pulse. The compensated chirp had an increasing frequency pattern to cancel out the cochlear delay. An enhanced chirp had a delay pattern that enhances the assumed cochlear delay. The pulse had a usual cochlear delay at the auditory peripheral. As a result, the accuracy of synchrony judgment was the highest in the pulse and higher in the enhanced chirp than the compensated chirp, implying that there is an asymmetric aspect. The purpose of this study is to investigate how our auditory system processes this asymmetric aspect, and what amount of temporal collapse was tolerated. We also measured the accuracy of synchrony judgment using stimuli that reverse the cochlear delay (the higher-frequency components arrive behind the lower-frequency components), or enhance the delay of lower-frequency components up to 8 times.

3aPP29. Do cochlear mechanisms explain the noise-disruption of the auditory brainstem response to speech? Helen E. Nuttall, Antje Heinrich, David R. Moore, and Jessica de Boer (MRC Inst. of Hearing Res., Science Rd., University Park, Nottingham NG7 2RD, United Kingdom, helen@ihr.mrc.ac.uk)

In background noise, the timing precision of the auditory brainstem response to speech (speech-ABR) is disrupted and the response latency increases. The severity of the disruption has been correlated with listeners'

ability to understand speech-in-noise. To date, although a central mechanism is assumed, the locus of the speech-ABR timing disruption is not clear. The present study aimed to investigate the contribution of different cochlear mechanisms to noise-induced latency increases. A first experiment examined the "cochlear place" mechanism, by which the latency of the response increases as cochlear origin moves towards lower frequency regions. The results showed that the speech-ABR reflects an average over responses from a broad range of cochlear regions, which respond with substantial relative delays. This implies that cochlear place can potentially have large effects on masked speech-ABR latency. Another mechanism that is known to be involved in noise-induced ABR latency increases is neural adaptation. This is presumed to occur at the inner hair cell-nerve junction and is thought to reflect cochlear masking. Thus, if this mechanism contributes to speech-ABR latency increases in noise, we would expect this contribution to depend on cochlear frequency selectivity and amplification gain. This hypothesis is tested in the second experiment.

3aPP30. Modeling human auditory evoked brainstem responses to speech syllables. James Harte (Inst. of Digital Healthcare, Univ. of Warwick, International Digital Lab., Coventry CV47AL, United Kingdom, harte_j@wmg.warwick.ac.uk), Filip M. Roenne (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, Copenhagen, Denmark), and Torsten Dau (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, Lyngby, Denmark)

Auditory evoked brainstem responses (ABR) are routinely used to assess the neural encoding of sound. Various types of stimuli have been historically considered, and there is a current increasing trend towards the use of syllables, speech and complex (non-speech) sounds. Despite the peripheral origin of ABRs, the nonlinear processing within the cochlea and brainstem makes interpreting responses to these new stimuli a challenge. A recent model was developed [Rønne *et al*. (2012)] to simulate ABRs to transient sounds such as tone pulses, clicks and rising chirps as a function of stimulus level. The present study extends this model to simulate synthetic /ba/, /da/ and /ga/ syllable-evoked ABRs, where the stimuli only differ in their spectral energy of the second formant, f2, within their first 60 ms. The model takes the convolution of the instantaneous discharge rates using a "humanized" nonlinear auditory-nerve model of Zilany *et al*. (2009) and an empirically derived unitary response function, assumed to reflect contributions from different cell populations within the auditory brainstem. The ABR model was used to explore the physiological basis and spectro-temporal characteristics of key features observed in syllable-evoked ABRs in the literature [Skoe *et al*. (2011), Hornickel *et al*. (2009)].

3aPP31. Sensitivity to stimulus polarity in speech-evoked frequency-following responses. Steve J. Aiken (School of Human Commun. Disord., Dalhousie Univ., 1256 Barrington St., Box 15000, Halifax, NS B3H 4R2, Canada, steve.aiken@dal.ca) and David W. Purcell (National Ctr. for Audiol., Western Univ., London, ON, Canada)

It has been suggested that frequency-following responses recorded to speech sounds presented in opposite polarities can be added to emphasize responses related to the periodicity envelope (e.g., at the fundamental frequency, f0) or subtracted to emphasize responses related to the stimulus fine-structure (e.g., harmonics near formant peaks) because inverting stimulus polarity has little effect on the stimulus envelope. This hypothesis was tested by comparing frequency-following responses to several tokens of two vowels (/a/ and /i/) presented twice in one polarity and once in the opposite polarity, from nine normal-hearing listeners. At harmonics near formant peaks, most listeners displayed frequency-following responses closely related to stimulus fine-structure (i.e., responses that followed stimulus polarity). However, response amplitude and phase at f0 varied greatly across polarities for many listeners, suggesting that the f0 response does not reflect a simple encoding of the periodicity envelope or the fine-structure of the first harmonic. Most listeners had similar responses when identical stimuli were presented in the same polarity, so these varied polarity-sensitive responses were not likely related to temporal encoding difficulties or background electrophysiological noise. The summed alternating polarity frequency-following response at f0 might reflect encoding of both speech fine-structure and the periodicity envelope.

3aPP32. Auditory evoked responses to a frequency glide following a static pure tone. Wen-Jie Wang (Speech-Language-Hearing Sci., Grad. Ctr., CUNY, 365 Fifth Ave., New York, NY 10016, wwang2@gc.cuny.edu), Chin-Tuan Tan (Dept. of Otolaryngol., School of Medicine, New York Univ., New York, NY), and Brett A. Martin (Speech-Lang.-Hearing Sci., Grad. Ctr., CUNY, New York, NY)

In this study, we look at the auditory evoked response to a frequency glide following a static pure tone. A frequency glide is a frequency ramp with specific frequency change range (Δf) and duration (Δt). Frequency change rate (Δf/ Δt) and direction (increasing or decreasing frequency) of a glide are important cues for speech perception. P1-N1-P2 acoustic change complex (ACC) responses to increasing or decreasing frequency glides were observed in the recordings of normal hearing subjects. Subjects were also asked to behaviorally discriminate similar stimuli with a fixed Δt at 50 ms or 200 ms and a varying Δt in a separate experiment. Similar findings were obtained with glides at both 500 and 1 kHz base frequency. In these preliminary data, we observed larger N1-P2 responses with the glides of fixed Δt 50 ms at both 500 Hz and 1000 Hz base frequency. However, larger N1-P2 responses for increasing glides than for decreasing glides were only observed with glides at 500 Hz base frequency. Larger N1-P2 response at shorter Δt seems to tally with the smaller behavioral threshold of Δt difference between stimulus with a fixed Δt at 50 ms and stimulus with varying Δt.

3aPP33. Evidence for modulation rate specific adaptation in the frequency following response? Hedwig E. Gockel, Alexandra Krugliak (MRC-Cognition and Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom, hedwig.gockel@mrc-cbu.cam.ac.uk), Christopher J. Plack (Audiol. and Deafness Res. Group, Univ. of Manchester, Manchester, United Kingdom), and Robert P. Carlyon (MRC-Cognition and Brain Sci. Unit, Cambridge, United Kingdom)

We used an adaptation paradigm to investigate whether the frequency following response (FFR) would show evidence for neurons tuned to modulation rate in humans, as has been previously shown in the inferior colliculus of the macaque using fMRI [Baumann *et al*. Nat. Neurosci. **14**, 423–425 (2011)]. The FFR to a 100-ms, 75-dB SPL, target complex tone with an envelope rate of 213 Hz was measured for ten subjects. The target was preceded by a 200-ms, 75-dB SPL, adaptor complex with an envelope rate of 90, 213, or 504 Hz. All complexes contained alternating-phase harmonics from approximately 3.9 to 5.4 kHz. A "vertical" montage (+ Fz, - C7, ground = mid-forehead) was used, for which the FFR is assumed to reflect phase-locked neural activity from generators in the rostral brainstem. The results showed significant adaptation effects in the spectral magnitude of the 213-Hz envelope-related component of the FFR. However, the identical-rate adaptor did not generally produce more adaptation than the different-rate adaptors. Hence, the present results do not provide evidence for neurons tuned to modulation rate in the human brainstem. [Work supported by Wellcome Trust Grant 088263.]

3aPP34. Simultaneously evoked auditory potentials: A novel paradigm for measuring auditory-evoked electroencephalographic activity at successive levels of the auditory neuraxis. Christopher Slugocki, Dan J. Bosnyak, and Laurel J. Trainor (Psychology, Neurosci., & Behaviour, McMaster Univ., 1280 Main St. West, Hamilton, ON L8S4L8, Canada, slugocc@gmail.com)

Recent electrophysiological work has evinced a capacity for plasticity in subcortical auditory nuclei in adults [Skoe and Kraus (2010)]. Similar plastic effects have been measured in cortically generated auditory potentials [e.g., Bosnyak *et al*. (2007); Näätänen (2008)], but it is unclear how the two interact. Here we present simultaneously evoked auditory potentials (SEAP), a novel paradigm designed to concurrently elicit electrophysiological brain potentials (EEG) from inferior colliculus (IC), thalamus, and primary and secondary auditory cortices. We use a specially designed stimulus consisting of a carrier frequency (500 Hz), amplitude-modulated at the sum of 37 and 81 Hz (depth 100%). We have shown that it elicits a 500 Hz frequency-following response (FFR; generated in IC), 81 (subcortical), and 37 (primary auditory cortex) Hz steady state responses, and mismatch negativity (when there is an occasional change in carrier frequency; secondary auditory cortex). Furthermore, cortical and subcortical processes are linked as the amplitude of the FFR predicts the amplitudes of the 37 and 81 Hz responses. SEAP offers a new window into the dynamics of encoding along the auditory pathway as well as a new and inexpensive tool with which to measure plasticity at multiple levels of the auditory neuraxis in individuals.

3aPP35. Measuring subcortical and cortical neural activities for music perception: A multilevel electroencephalography study. Inyong Choi, Scott Bressler, and Barbara G. Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, iychoi@bu.edu)

Music perception requires the activation and coordination of many neuronal centers and pathways of the peripheral and central auditory pathway, which are highly overlapped with those of speech communication. Previous experiments have shown influences of musical training on brainstem encoding in the auditory periphery and long-term plasticity in the cortex. By testing listeners with different musical experience we hope to better to understand differences in central auditory processing across individual listeners. Here we explore methods for measuring subcortical and cortical neural activity in response to musically relevant stimuli by electroencephalography (EEG). A passive mismatch negativity (MMN) paradigm using familiar musical intervals was presented to subjects to measure late evoked potentials in response to deviations in absolute musical interval and musical consonance. Brainstem frequency following responses (FFRs) for carrier frequencies and phase locking values to the beat-related envelopes were simultaneously measured. Results from these experiments can provide a means to objectively quantify individual differences in central auditory processing related to musical ability through non-invasive, electrophysiological methods.

WEDNESDAY MORNING, 5 JUNE 2013 — 512CG, 9:00 A.M. TO 11:00 A.M.

Session 3aSA

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

Yun Jing, Cochair
Mech. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695

Dean Capone, Cochair
Penn State, P.O. Box 30, State College, PA

Invited Papers

9:00

3aSA1. Interactive behavior of internal resonators in acoustic metamaterials under impact pulse loading. Kwek Tze Tan and C. T. Sun (Aeronautics and Astronautics, Purdue Univ., 701 West Stadium Ave., West Lafayette, IN 47907, kttan@purdue.edu)

Acoustic metamaterials exhibit negative effective mass density when the lattice system consists of mass-in-mass microstructural units. It is found out that the effective mass density becomes frequency dependent and displays negativity for frequencies near the resonant frequency of the internal resonators. The effect of a negative mass property implies that stress wave propagation is prohibited; leading to structural applications like vibration control, impact protection, and shock wave mitigation. Under impact loading, internal resonators are revealed to effectively reduce the displacement/velocity of the overall structure, and attenuate a specifically designed range of frequency where the negative effective mass density is exhibited. However, researchers have yet to study the mutual interaction between the internal resonators. Knowing how adjacent resonators interact and response to dynamic profile of preceding resonators may lead to more efficient design for stress wave attenuation. In this paper, we performed detailed investigation on the interactive behavior of internal resonators in acoustic metamaterials under an impact pulse load. Finite element analysis results show that when internal resonators are adjacently placed, they produce a coupled resonance effect, resulting in a leakage of frequency just below the resonant frequency of the resonators. This frequency leakage can lead to energy storage and harvesting applications.

9:20

3aSA2. Acoustic supercoupling and enhancement of nonlinearities in density-near-zero metamaterial channels. Caleb F. Sieck (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Romain Fleury (Dept. of Elec. & Comput. Eng., The Univ. of Texas at Austin, Austin, TX), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu), and Andrea Alù (Dept. of Elec. & Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

Recent theoretical and experimental work has demonstrated that acoustic wave tunneling and energy squeezing can be achieved using density-near-zero (DNZ) metamaterial channels [Fleury *et al*, J. Acoust. Soc. Am. **132**(3), 1956 (2012)]. These channels are directly analogous to supercoupling of electromagnetic waves in near-zero permittivity channels. In optics, the field enhancement and uniformity of response within a near-zero permittivity channel can be employed to produce switching behavior, harmonic generation, and wave mixing even with low amplitude input intensities. These optical channels have been already shown to significantly outperform enhancement of nonlinearity in conventional Fabry-Pérot resonant gratings [Argyropoulos *et al*., Phys. Rev. B **85**, 045129 (2012)]. The analogous properties of velocity field within a DNZ metamaterial channel can result in significant and uniform amplification that may be employed to enhance material or structural nonlinearities in the channel for applications like transmission switches. This work presents recent analytical and finite element modeling of the use of DNZ channels to enhance acoustic nonlinearities. It also explores and discusses metamaterial mechanisms for attaining a tailored and enhanced nonlinear response.

9:40

3aSA3. Acoustic double negativity with coupled-membrane metamaterial. Guancong Ma, Min Yang, Zhiyu Yang, and Ping Sheng (Dept. of Phys., Hong Kong Univ. of Sci. and Technol., HKUST, Clear Water Bay, Kowloon 123456, Hong Kong, phmgc@ust.hk)

Over the past decade, the emergence of acoustic metamaterials has considerably broadened the possibility of acoustic wave manipulations. Within this area, exotic effective constitutive parameters (mass density and bulk modulus) are a most hotly pursued topic, at the core of which is the realization of acoustic double negativity. Here, we show with experiments, simulations and homogenization that a single resonant structure can achieve acoustic double negativity. The metamaterial is comprised of two decorated elastic membranes, which are connected together by a rigid ring. Impedance measurement reveals that the system's transport behavior is governed by the three eigenmode resonances in the sub-kHz regime, which are separately tunable via system parameters. Measured displacement profiles at the sample surfaces using laser vibrometer show that the system's eigenmodes are, respectively, dipolar or monopolar in their nature. The simplicity of the metamaterial also enables us to retrieve its effective mass density and effective bulk modulus by performing homogenization. The results help explaining the physics behind the transport properties, and confirm that a double-negative passband is realized in a frequency range (around 500–800 Hz). Excellent agreement between experiments, simulations, and theory is achieved.

10:00

3aSA4. Conceal an acoustic sensor by using single-negative acoustic materials. Jianchun Cheng, Bin Liang, and Xuefeng Zhu (Inst. of Acoust., Dept. of Phys., Nanjing Univ., 22 Hankou Rd., Nanjing, Jiangsu, Nanjing 210093, China, jccheng@nju.edu.cn)

A coordinate transformation scheme is proposed to make a sensor acoustically undetectable while allowing it to receive external information. The designed structure only comprises complementary media (CM) whose acoustic parameters are single-negative rather than double-negative, and are totally independent of those of the sensor and the background medium. The numerical results show that the incident wave can pass across the CM changelessly, and it is therefore reasonable to identify such a structure as an acoustic cloak which enables one to "hear without being heard." Further, a nonlinear transformation scheme is developed, in an attempt to simplify the practical realization to the fullest extent. The designed multi-layered structure is formed by alternately arranging one single pair of CM with homogeneous isotropic single-negative parameters which are also totally independent of the sensor and background medium. The numerical results show that acoustic scattering from the sensor is suppressed considerably when the number of bilayers is large enough. The feasibility of designing acoustic cloaks by single-negative media and the flexibility in choosing the acoustical parameters of cloak may significantly facilitate the experimental realization of acoustic cloaks and promise new possibilities in a wide range of acoustics, optics, and engineering applications.

Contributed Papers

10:20

3aSA5. Experimental validation of the band-gap and dispersive bulk modulus behavior of locally resonant acoustic metamaterials. Matthew Reynolds, Yan Gao, and Steve Daley (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, mjr304@soton.ac.uk)

Over the last decade there has been significant interest in the design and production of acoustic metamaterials with physical qualities not seen in naturally occurring media. Progress in this area has been stimulated by the desire to create materials that exhibit novel behavior such as negative refraction due to negative material parameters, and band gaps in the frequency response of the material. An acoustic metamaterial is presented that consists of an acoustically transparent mesh with an array of split hollow spheres. Split hollow spheres are analogous to the split ring resonators found in many electromagnetic metamaterials and act as Helmholtz resonators providing a resonant band gap at low frequencies where achieving a Bragg gap would be impractical, and providing a dispersive effective bulk modulus that can become negative. Since an eventual goal of the work is to produce such materials on a micro-scale, the metamaterial is designed for, and produced using, 3D printing techniques (additive layer manufacturing). Results are presented for material comparing theory and experiment, and methods for increasing the bandwidth of the behavior in question are proposed, including a mixed resonator solution and the integration of active components into the material.

10:40

3aSA6. A meta-mass mechanical energy valve and battery—A key to vibration energy harvesting. Joh J. McCoy (The Catholic Univ. of America, Michigan Ave., Washington, DC 20064, mccoy@cua.edu)

A meta-mass comprised of a rigid housing element densely filled by a multiplicity of oscillators, is designed to act, when attached to a vibrating structure, as a one-way energy valve and mechanical battery. When the oscillators are joined with mechanical/electrical converters, the meta-mass becomes a vibration energy harvester that has a capacity that far exceeds harvesters comprised of a limited number of Newtonian mass elements. When viewed from the structure side of its attachment, the meta-mass device action as a one-way valve appears as an energy sink, such that when sufficiently strong initiates a transport process drawing energy from remote locations, thereby solving a fundamental aperture problem to energy harvesting, occasioned by the spatial diffuseness of the energy source. When viewed from the device side of the attachment, the meta-mass device action as a battery provides for the accumulation of mechanical energy, thereby solving a second fundamental problem to energy harvesting, occasioned by the arrival of the vibration energy as a sequence of weak packets. A meta-mass vibration energy harvester (MMVEH) is contrasted with the present art of vibration energy harvesters (VEH).

Session 3aSCa

Speech Communication: Imitation, Accommodation, and Convergence in Speech Communication

Molly E. Babel, Cochair
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Kuniko Nielsen, Cochair
Linguistics, Oakland Univ., 320 O'Dowd Hall, Rochester, MI 48309-4401

Chair's Introduction—8:55

Invited Papers

9:00

3aSCa1. The cognitive basis of spontaneous imitation: Evidence from the visual world. Stephen D. Goldinger (Psychology, Arizona State Univ., 950 S. McAllister, Box 871104, Tempe, AZ 95287-1104, goldinger@asu.edu)

It is well-established that, when people are asked to identify and quickly repeat spoken words, they show a strong tendency to spontaneously imitate the vocal and/or phonetic characteristics of the stimulus tokens. There is mixed evidence, however, regarding the underlying basis of such imitation: Does it only reflect gestural attunement (as in Direct Realism), or does it also reflect cognitive principles of word perception and memory? The gestural attunement view has face validity, as imitation seems to require tacit knowledge of other peoples' articulatory actions. The role of memory is less obvious, although people can certainly imitate others from memory. In this talk, I will present evidence from three new experiments, pairing procedures from the "visual world" paradigm with a speech production task. Across studies, there is clear evidence that degrees of speech imitation are tightly connected to attention and memory processes that were engaged during initial exposure to spoken words. The results show clear imitation (in naming depicted objects), both with and without spoken words prompting responses, and show strong effects of competition among visual objects: Imitation increases when other potential objects have similar names, or even similar appearances. Spontaneous imitation is both a gestural and a cognitive behavior.

9:20

3aSCa2. Phonetic convergence is not a consequence of stimulus-response compatibility. Holger Mitterer (Dept. of Cognit. Sci., Univ. of Malta, University Ring Rd., Msida MSD2080, Malta, holgermitterer@yahoo.co.uk)

Moving to a different region within a country/language area tends to have the same consequences all over the globe. Phonetic patterns change over time, and they change in the direction of the pattern in the environment [Harrington *et al*. (2000), Sancier and Fowler (1997)]. This has often been attributed to a basic stimulus-response compatibility on a gestural level [Galantucci *et al*. (2009)]. The experimental evidence for a gestural stimulus-response compatibility, however, invariably confounds gestural with phonological compatibility, that is, gesturally incompatible distractor stimuli are also from a different phonological category [Mitterer and Ernestus (2008)]. In this presentation, I will present three lines of evidence that question a gestural stimulus-response compatibility driving phonetic convergence. A first experiment shows that phonetic convergence does not necessarily target concrete phonetic detail, but rather more global and abstract parameters. Second, I will show that there is no relationship between the amount of phonetic stimulus-response overlap and response latency, the typical measure of stimulus-response-compatibility effects. Finally, I will present data that show that two participants in a conversation can maintain a marked phonetic difference. The combined evidence indicates that phonetic convergence is not driven by a gestural stimulus-response compatibility.

Contributed Papers

9:40

3aSCa3. Evidence for an articulatory component of phonetic convergence from dual electromagnetic articulometer observation of interacting talkers. Mark Tiede (Haskins Labs., 300 George St., New Haven, CT 06511, tiede@haskins.yale.edu) and Christine Mooshammer (U.S.C. & Haskins Labs., Los Angeles, CA)

Speech audio and articulatory movements of age- and gender-matched speaker pairs have been recorded in face-to-face interaction using two electromagnetic articulometer (EMA) systems simultaneously. Speakers matched for language (AE:AE) were compared with mixed language backgrounds (AE:Spanish, AE:German). Tasks included synchronized, imitated, and spontaneous speech interspersed with repeated baseline utterances for evaluation of mutual accommodation. Euclidean distances (EDs) between vowel midpoint formant frequencies from the initial and final baseline tasks showed symmetric reductions associated with convergence which were larger for the speakers with the same language background. To test whether the observed reduction might be due to fatigue or repetition effects EDs were also calculated between AE speakers who did not participate in the same experiment. Because reduction was minimal for this comparison the possibility that convergence can be attributed to simple optimization is excluded. Kinematic and dynamic convergence was examined on coda velar gestures from the baseline tasks. EDs in peak velocity, sensor path distance, gesture amplitude, and stiffness computed between speakers all showed

decreases consistent with convergence. Associated within-speaker differences showed that accommodation was effectively symmetric, and because the speakers moved towards one another the difference was not due to practice or fatigue effects. [Work supported by NIH.]

10:00

3aSCa4. Visual enhancement of alignment to noisy speech. James W. Dias and Lawrence D. Rosenblum (Psychology, Univ. of California, Riverside, 900 University Ave., Riverside, CA 92521, jdias001@ucr.edu)

Talkers imitate aspects of perceived speech; a phenomenon known as speech alignment. Previous studies have found that talkers will align not only to auditory speech, but also visual speech information. Furthermore, talkers who interact in full view of a conversational partner will align more than talkers who interact with a partner they can only hear [Dias and Rosenblum, Perception **40**, 1457–1466 (2011)]. The purpose of the current investigation is to evaluate whether visual speech information increases speech alignment when the auditory speech is noisy. Participants shadowed (said out-loud) the utterances of a pre-recorded model presented under audio-alone or audiovisual conditions, either in noise (+10 db SNR) or in the clear. These shadowing tokens were then rated for their similarity (alignment) to the model's tokens using a matching task performed by naïve raters. Despite word identification performance being the same across all conditions, participants aligned more to the model when listening to speech in the clear than speech in noise. Further, the addition of visual information enhanced alignment for words presented in noise, but not for words presented in the clear. These data suggest the availability of visual speech information can enhance speech alignment when shadowing in noise.

10:20–10:40 Break

Invited Papers

10:40

3aSCa5. Reconciling diverse findings in studies of phonetic convergence. Jennifer Pardo (Psychology, Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, pardo@optonline.net)

Phonetic convergence occurs both when individuals interact in conversation and when listeners rapidly repeat words presented over headphones. Results from multiple studies examining phonetic convergence offer an array of often confusing and disparate findings. Reconciling such diverse findings is difficult without a clear rationale for engaging in one acoustic measure over another. The current paper proposes a paradigm that models perceptual and acoustic measures together. Measures of multiple acoustic-phonetic attributes were compared with a perceptual measure of phonetic convergence in a shadowing study. Although convergence was not significant in any acoustic measure alone, the combination of acoustic attributes predicted perceived phonetic convergence on an item-by-item basis. Because perceptual measures integrate across multiple acoustic-phonetic dimensions, future studies of phonetic convergence should use perceptual tasks to calibrate the relative contribution of individual acoustic-phonetic parameters.

11:00

3aSCa6. Phonetic convergence, communicative efficiency, and language distance. Ann Bradlow, Midam Kim (Linguistics, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, abradlow@northwestern.edu), and Minyoung Kim (Management and Int. Business, The Univ. of Kansas, Lawrence, KS)

Many English conversations across the globe today involve talkers with different language experiences. Here we show that, while language barriers challenge communicative efficiency, the detrimental effect of language distance may be mitigated by phonetic convergence. We analyzed a corpus of 42 conversations in which talker pairs solved a spot-the-difference puzzle by verbally comparing two scenes only one of which was visible to each talker ("diapix" task). Language distance was varied by pairing talkers who either matched or mismatched in language background and in native/nonnative status relative to the target language. Communicative efficiency was measured by task-completion-time and word type-to-token ratio. Phonetic convergence was assessed by perceptual similarity tests in which listeners ($n = 161$) compared samples from one talker's speech to samples from his/her partner's speech from either early or late portions of the conversation. In this test, greater similarity for late than early samples indicates convergence. Results showed a negative correlation between language distance and communicative efficiency, a negative correlation between language distance and phonetic convergence, and a mitigating effect of phonetic convergence on the negative correlation between language distance and communicative efficiency. This suggests that convergence may be an effective mechanism for overcoming the detrimental effects of a language barrier.

Contributed Paper

11:20

3aSCa7. Phonetic imitation by individuals with Autism Spectrum Disorders: Investigating the role of procedural and declarative memory. Jeff Mielke (Dept. of English, North Carolina State Univ., Tompkins 286, Raleigh, NC 27695-8105, jimielke@ncsu.edu), Kuniko Nielsen (Linguist Dept., Oakland Univ., Rochester, MI), and Lyra Magloughlin (Dept. of Linguist, Univ. of Ottawa, Ottawa, ON, Canada)

This study investigates the role of procedural and declarative memory in phonetic imitation, by examining the word- and phoneme-specificity of imitation produced by individuals with autism spectrum disorders (ASD). Previous research has shown that individuals with ASD process language differently from the Neurotypical population [e.g., Ullman (2004), Walenski *et al*. (2006)], with Autistic individuals relying more on declarative memory. Previous work with the general population has shown a robust effect of phonetic convergence [e.g., Pardo (2006)], as well as generalization and weak word-specificity effects [Nielsen (2011)]. To test whether individuals with ASD exhibit increased specificity, we used Nielsen's (2011) experimental paradigm, which has been shown to elicit generalized phonetic imitation in the general population. A linear mixed effects regression analysis revealed that increased VOT on the modeled phoneme /p/ was imitated by both ASD and control groups [$p < 0.05$]. However, different patterns emerged in phoneme-level specificity: the control group exhibited sub-phonemic generalization (increasing VOT on /p/ and /k/), while the ASD group exhibited a phoneme-specific pattern (increasing VOT only on /p/) [$p < 0.05$]. Furthermore, a stronger trend toward word-specificity was observed within the ASD group. Taken together, these results confirm the earlier finding that ASD individuals exhibit greater reliance on declarative memory.

11:40–12:10 Panel Discussion

Session 3aSCb

Speech Communication: Components of Informational Masking

Gaston Hilkhuysen, Cochair
Laboratoire de Mécanique et d'Acoustique, CNRS, 31, Chemin Joseph Aiguier, Marseille 13402, France

Yoshitaka Nakajima, Cochair
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Chair's Introduction—8:55

Invited Papers

9:00

3aSCb1. Understanding informational masking from a neural perspective. Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu)

Historically, psychoacousticians have divided the influence of a task-irrelevant sound (a masker) on perception of a task-relevant sound (the target) into components of (1) energetic masking and (2) informational masking. In this apportionment, energetic masking is defined as that masking that can be accounted for by considering how reliably the target is represented in the auditory periphery, and how much the masker disrupts this target representation. In contrast, the term informational masking is a catchall representing any effects of the masker that could not be accounted for by energetic masking. This talk presents a framework for understanding informational masking from a neural perspective, building on both behavioral results and neuro-imaging data. In this account, informational masking is a result of bottlenecks in the neural processing of acoustic information, a problem that the brain mediates by engaging auditory attention. Auditory attention operates by modulating the representation of different auditory objects making up a particular acoustic scene, resulting in a relative enhancement of whatever object is in the attentional foreground at the expense of the representation of competing sources. Thus, most informational masking arises from failures of target formation and/or failures of target selection.

9:20

3aSCb2. The segregation of target speech from competing speech when listening in a second language. Bruce A. Schneider (Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, bruce.schneider@utoronto.ca)

Introducing a perceived spatial separation (via the precedence effect) between target speech and competing speech reduces the signal-to-noise ratio (SNR) required for the recognition of key target words by 3–7 dB in syntactically correct but semantically anomalous target sentences such a "A rose could paint a fish" (target words: rose, paint, fish). A recent study of monolingual Mandarin participants listening to anomalous sentences in Mandarin Chinese suggests that perceived spatial separation releases listeners from informational masking (IM) by facilitating access to the lexicon at the morphemic level, presumably by enhancing the segregation of the target speech from the competing speech background. This raises the interesting question as to whether perceived spatial separation facilitates lexical access in the same way for people listening in the second language (L2) as it does when listening in their first language (L1) since access to the L2 lexicon is likely to be less robust than access to the L1 lexicon. A second interesting question is the extent to which operating in a L2 environment affects release from IM when the competing speech is in L1. These issues will be addressed in L2 English listeners whose L1 is either Chinese or Korean.

9:40

3aSCb3. Exploring the factors predictive of informational masking in a speech recognition task. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Med. Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov) and Anna C. Diedesch (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

The effects of informational masking (IM) can be recast as a question of which cues to sound source identity (auditory object formation) are most useful for overcoming IM. We hypothesize that individual differences are related to specific interactions of stimulus and listener-specific variables that determine the effectiveness of the auditory object formation process. Results from our laboratory generally support the well-established relationship between performance and stimulus variables such as spectrotemporal cues (in this case, voice differences) and spatial cues (talker locations). In addition, the listener-specific variables of age and hearing loss were found to interact with the stimulus variables and to be correlated with potential mediating variables such as interaural time sensitivity and minimum levels at which speech identification was possible. Future work will involve developing predictive models that focus on identifying the mediating variables responsible for increased susceptibility to IM and efficient tests to reveal these relationships in individual listeners. The clinical relevance of the ability to identify factors predictive of IM susceptibility will be discussed, including the potential for improved fitting of hearing aids and cochlear implants.

10:00

3aSCb4. Interactions between listening effort and masker type on the energetic and informational masking of speech stimuli.
Douglas Brungart (Audiol. and Speech Ctr., Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.brungart@us.army.mil), Nandini Iyer, Eric Thompson, Brian D. Simpson (711th HPW, Air Force Res. Lab., Wright-Patterson AFB, OH), Sandra Gordon-Salant, Jaclyn Shurman, Chelsea Vogel (Dept. of Hearing & Speech Sci., Univ. of Maryland, College Park, MD), and Kenneth W. Grant (Audiol. and Speech Ctr., Walter Reed NMMC, Bethesda, MD)

In most cases, normal-hearing listeners perform better when a target speech signal is masked by a single irrelevant speech masker than they do with a noise masker at an equivalent signal-to-noise ratio (SNR). However, this relative advantage for segregating target speech from a speech masker versus a noise masker may not come without a cost: segregating speech from speech may require the allocation of additional cognitive resources that are not required to segregate speech from noise. The cognitive resources required to extract a target speech signal from different backgrounds can be assessed by varying the complexity of the listening task. Examples include: (1) contrasting the difference between the detection of a speech signal and the correct identification of its contents; (2) contrasting the difference between single-task diotic and dual-task dichotic listening tasks; and (3) contrasting the difference between standard listening tasks and one-back tasks where listeners must keep one response in memory during each stimulus presentation. By examining performance with different kinds of maskers in tasks with different levels of complexity, we can start to determine the impact that the informational and energetic components of masking have on the listening effort required to understand speech in complex environments.

WEDNESDAY MORNING, 5 JUNE 2013 — 510A, 8:55 A.M. TO 12:00 NOON

Session 3aSP

Signal Processing in Acoustics and ASA Committee on Standards: Methods and Applications of Time-Frequency Analysis

Leon Cohen, Cochair
City Univ. of New York, Hunter-Physics, 695 Park. Ave., New York, NY 10065

Patrick J. Loughlin, Cochair
Bioengineering, Univ. of Pittsburgh, 302 Benedum Hall, Pittsburgh, PA 15261

Chair's Introduction—8:55

Invited Papers

9:00

3aSP1. Speaker identification made easy with pruned reassigned spectrograms.
Sean A. Fulop (Linguistics, California State Univ. Fresno, 5245 N Backer Ave., PB92, Fresno, CA 93740-8001, sfulop@csufresno.edu) and Youngwook Kim (Elec. and Comput. Eng., California State Univ. Fresno, Fresno, CA)

One common scenario for speaker identification presents the task of identifying samples of speech from members of a previously enrolled group. One recent (and typical) set of results used 36 s of speech from each speaker to train Gaussian models by expectation-maximization during enrollment, and 20 s of speech for the test samples. Three major problems with this procedure are (1) sensitivity to noise; (2) impractical amounts of speech are required; (3) computationally expensive training is required. In our study, the reassigned spectrogram is pruned using phase-derivative indicator functions to provide a sparse time-frequency matrix of very small (40 ms) samples of speech. The pruning eliminates Gaussian noise up to ~1 dB SNR. Principle components analysis provided a set of 30 features from each spectrogram. Using an enrolled group of 24 speakers recorded under low-fidelity conditions, 83% identification accuracy (comparable to state of the art results with 6 dB SNR) was achieved from real-time classification methods (e.g., support vector machines) without need for extensive training. Moreover, these results extend to 1 dB SNR where standard techniques break down. The three main problems of speaker identification can thus be better addressed by our methodology.

9:20

3aSP2. Phase-space equation for wave equations.
Leon Cohen (City Univ. of New York, Hunter-Physics, 695 Park. Ave., New York, NY 10065, leon.cohen@hunter.cuny.edu)

Transforming a space-time function into the phase space of position and wave number offers considerable insight into the nature of the function, and also has many practical applications. If the function is governed by a wave equation then the common procedure is to solve the wave equation and then calculate the phase space distribution function for the solution. We show that significant advantages ensue if one transforms the original differential equation into a phase space differential equation. We give a number of examples and show that phase space equations are often more revealing than the original equation and lead to new approximation methods. [Work supported by ONR.]

9:40

3aSP3. Nonstationary vibration analysis in the smoothed Wigner domain. Lorenzo Galleani (Politecnico di Torino, Corso Duca degli Abruzzi 24, Torino 10129, Italy, galleani@polito.it)

Dynamical systems represent fundamental models for vibration analysis. When the input of such dynamical systems is nonstationary also the output is nonstationary, and its frequency content changes with time. Time-frequency analysis provides an effective representation of this time-varying spectrum. Even more effective is the direct time-frequency representation of dynamical systems. We first show how to transform a dynamical system in the domain of the smoothed Wigner distribution. The result is a time-frequency dynamical system whose input and output are the smoothed Wigner distributions of the input and output signals in time, respectively. Then, we illustrate how to compute the smoothed Wigner output when the input to the dynamical system in time belongs to a class of common nonstationary inputs, including a delta function and a short duration sinusoid. Finally, we show how to obtain the smoothed Wigner output when the input is a linear combination of these nonstationary signals. We provide a series of examples that show how the time-frequency representation of dynamical systems can unveil the spectral structure of nonstationary vibrations.

10:00

3aSP4. Time-frequency analysis with Bayesian filtering methods for dispersion tracking and geoacoustic inversion in the ocean. Nattapol Aunsri and Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd, Newark, NJ 07102, michalop@njit.edu)

Iterative and sequential Bayesian filtering methods have been previously used in dispersion tracking for long range sound propagation in oceanic environments. The methods rely on accurate modeling of (i) the acoustic field in the frequency domain as a function of time and (ii) the statistical model governing the behavior of errors in the measured data. Normal distributions are typically used for the latter but may not necessarily be the most accurate models for the description of the data. We investigate alternative methods for describing the statistical errors in power spectra, which we then use for linking the extracted time-frequency information to geoacoustic inversion. Probability density functions computed for frequencies and arrival times via filtering are propagated "backwards" through sound propagation models for quantification of the uncertainty in the estimation process. [Work supported by ONR].

10:20

3aSP5. Single snapshot spatial array processing using time-frequency distributions. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Several signal processing applications, such as spatial beamforming, rely on data collected by an array of sensor to estimate to enhance a weak signal in the presence of noise (e.g., ambient noise or clutter). The commonly applied eigenstructure subspace methods to this signal denoising problem assume stationary signals and require multiple snapshots to correctly estimate the covariance matrix of the array data. However, these multiple snapshots and stationarity requirements can be hard to meet in practical scenarios involving among others a rapidly moving source (which causes differential Doppler effect among sensors) or a single snapshot of the aspect-dependent scattering of an unknown target as measured by a monostatic sonar system (such as side-scan sonar). To handle these scenarios, we propose to form a generalized space-time-frequency covariance matrix from the single-snapshot data by computing Cohen's class time-frequency distributions between all sensor data pairs. The eigenstructure of this space-time-frequency covariance matrix allows to enhance the localization of the signal of interest while spreading the noise power in the time-frequency domain. Hence, this approach is especially suited to handle nonstationary echoes of underwater target resonances that are highly localized in the time-frequency domain as demonstrated using numerical and at-sea data.

10:40

3aSP6. Time-frequency analysis of transient high-frequency dispersive guided waves on tilted cylindrical shells: Review. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu) and Scot F. Morse (Comput. Sci. Div., Western Oregon Univ., Monmouth, OR)

Measurements of the back scattering by bluntly truncated tilted cylindrical shells in water reveal a dependence on the aspect angle, which can be interpreted using geometrically described coupling mechanisms [Morse *et al*., J. Acoust. Soc. Am. **103**, 785–794 (1998)]. By exciting a shell with a suitable acoustic pulse localized in time, the recorded backscattering response reveals a significant evolution of the spectrum as a function of time and tilt angle. This evolution was interpreted using the dispersion relations of the relevant high-frequency shell guided waves [Morse and Marston, J. Acoust. Soc. Am. **111**, 1289–1294 (2002)]. The coupling conditions are affected by the mode properties. This interpretation was facilitated by also computing the scattering properties of an infinitely long titled shell [Morse and Marston, J. Acoust. Soc. Am. **106**, 2597–2600 (1999)] and by measuring and modeling the contributions of helical and meridional rays [Morse and Marston **112**, 1318–1326 (2002); Blonigon and Marston, **112**, 528–536 (2002)]. One of the shells investigated was also convenient for a quasi-holographic imaging of the bistatic scattering of short pulses [Baik *et al*., 130, 3838–3851 (2011)]. [Work supported by ONR.]

Contributed Papers

11:00

3aSP7. Time frequency analysis techniques for long range sediment tomography. Gopu R. Potty and James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu)

Long range sediment tomography inversion technique requires accurate estimation of the arrival times of acoustic normal modes. The inversion technique is based on minimizing the difference between the forward model predictions and the data using a global optimization scheme. Predictions are computed using a trial parameter set which is iteratively modified until the algorithm converges. The modal arrival times are calculated from the time-frequency analysis of broadband acoustic data collected on a single hydrophone. During the initial stages of the development of the inversion scheme, Fourier-based spectrograms and wavelet-based scalograms were used. Taking advantage of some of the recent developments, dispersion based short time Fourier transform (DSTFT) and warping transform techniques were used for the time-frequency analysis, in recent times. This paper will

summarize and compare the performance of these time frequency techniques in the context of long range sediment tomography. Data from some of the recent field tests (Shelfbreak Primer and Shallow Water-06 experiments) will be analyzed for this study. Finally, time-frequency analysis performance of another new technique, Modified S transform, will be examined using the field data. [Work supported by Office of Naval Research].

11:20

3aSP8. Time-frequency analysis of wood anomalies in acoustics. Jingfei Liu and Nico F. Declercq (George W Woodruff School of Mech. Eng., Georgia Inst. of Technol., 2, rue Marconi, Metz 57070, France, benjamin.jf.liu@gatech.edu)

When performing a Fourier transform on the reflection from a periodically corrugated surface for a normally incident beam the reflection spectrum is obtained. Steep dips have been observed from such reflection spectrum and are commonly called Wood anomaly. Because the frequencies at which Wood anomalies appears coincide with the frequencies where stimulation of Scholte-Stoneley waves is expected, Wood anomalies are traditionally explained as being caused by Scholte-Stoneley waves and as drain of energy. In this work, time-frequency domain analysis is made on the reflection from corrugated surfaces. The analysis suggests that Wood anomalies only occur when the entire reflection is processed by a Fourier transform, whereas for short time Fourier analysis this phenomenon does not seem to occur. The analysis also shows that energy is delayed in the time domain at the generation frequency of Scholte-Stoneley waves, and is therefore not necessarily drained. From the time-frequency point of view the source of Wood anomalies is re-examined and the actual relationship between the surface wave generation and Wood anomalies is also investigated.

11:40

3aSP9. Precise order tracking analysis of time-varying vibro-acoustic signature from rotating machines. Jin-Ho Bae (Mech. Eng., KAIST, 291 Daehak-ra, Yuseong-gu, Daejeon 305-701, South Korea, supurpower@kaist.ac.kr), Jeong-Guon Ih (Mech. Eng., KAIST, Daejeon, Chungcheongnam-da, South Korea), and Sang-Ryeol Kim (System Dynam., KIMM, Daejeon, Chungcheongnam-do, South Korea)

Transient operation of rotating machines usually exhibit a large variation in rotating speed and frequencies of interest often fluctuate due to RPM variations. Order tracking is useful for the transient analysis of vibro-acoustic signal from a rotating machine. A fine resolution is required both in time and frequency domains to trace the spectral magnitude of the interested frequency in precision. In this study, characteristics of popular order tracking methods using short time Fourier transform (STFT) and Vold-Kalman filter are studied in the viewpoint of calculation time and resolution. They are compared with a new tracking method using variable frequency resolution STFT (VFR-STFT), being suggested for the sound quality analysis. For a test signal, vehicle interior noise measured in an idle run-up condition is used. Under the same signal processing conditions, time and frequency domain resolutions produced by three methods are compared. It is observed that VFR-STFT detects bandwidth of high order components of engine firing frequency precisely and it produced the finest resolution in time and frequency domain among three methods. Although VFR-STFT needs a bit more calculation time compared to the other methods, it is useful for the cases that do not require a real time analysis.

WEDNESDAY MORNING, 5 JUNE 2013 511AD, 8:55 A.M. TO 9:40 A.M.

Session 3aUWa

Underwater Acoustics and Engineering Acoustics: Acoustic Sensing Via Fiber Optics

Jeremy Brown, Cochair
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Andrew R. McNeese, Cochair
ARLUT, 10000 Burnet Rd., Austin, TX 78758

Chair's Introduction—8:55

Invited Paper

9:00

3aUWa1. Tracking a human walker with a fiber optic distributed acoustic sensor. Emery M. Ku and Gregory L. Duckworth (OptaSense, 27 Moulton St., Cambridge, MA 02138, emery.ku@optasense.com)

Traditional ground sensors can be used as security systems to detect third party intrusion, but are costly and cumbersome to scale up to cover a large area. Fiber optic distributed acoustic sensing (DAS) offers a solution that economically and instantaneously monitors up to 40 linear km of a border or perimeter with a single system. The output of the OptaSense third generation fiber interrogation unit is spatially dense, coherent, and exhibits characteristics expected of traditional strain sensors. In this paper, DAS data are shown to indicate a walker's directionality and support beamforming algorithms for localization.

Contributed Paper

9:20

3aUWa2. Measurements of Brillouin gain spectra in erbium-doped optical fibers for long-distance distributed strain and temperature sensing. Mingjie Ding, Yosuke Mizuno, Neisei Hayashi, and Kentaro Nakamura (Precision and Intelligence Lab., Tokyo Inst. of Technol., R2-26, 4259 Nagatsuta-cho, Midori-ku, Yokohama, Kanagawa, Tokyo 226-8503, Japan, ding.m.aa@m.titech.ac.jp)

Brillouin scattering, one of the most important acousto-optical phenomena, occurs when lightwave interacts with periodical refractive-index change caused by the acoustic wave. The frequency downshift from the incident light to the backscattered Stokes light is called the Brillouin frequency shift (BFS), which depends on the strain/temperature applied to the optical fiber. Fiber-optic distributed Brillouin sensors have already been widely utilized to monitor strain/temperature conditions of structures including buildings, bridges and cables. The measurement range of the sensing system is partially limited by the optical propagation loss inside the sensing fiber. One solution is to employ special fibers capable of amplifying Brillouin signal. In this work, we employ erbium-doped fibers (EDFs) as sensing fibers, which are commonly used as 1550 nm optical amplifiers. As the first step, we investigated the strain/temperature dependence of Brillouin gain spectra (BGS) in EDFs with three different erbium concentrations. For all the samples, with increasing strain/temperature, the BGS shifted toward higher frequency, which was the same as that of silica fibers. Moreover, as the erbium concentration increased, the strain coefficient of BFS was increased, while its temperature coefficient was reduced. These results will definitely contribute to the future research of EDF-based distributed strain/temperature sensing systems.

WEDNESDAY MORNING, 5 JUNE 2013 — 511AD, 10:00 A.M. TO 11:40 A.M.

Session 3aUWb

Underwater Acoustics: Experimental Sensors and Methods

Jeremy Brown, Cochair
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Andrew R. McNeese, Cochair
ARLUT, 10000 Burnet Rd., Austin, TX 78758

Contributed Papers

10:00

3aUWb1. A towable combustive sound source for ocean surveys and ocean acoustics experiments. Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and David K. Wareham (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The Combustive Sound Source (CSS) is a broadband impulsive sound source that generates a wide bandwidth underwater acoustic signal, similar to explosives and airguns, yet allows for a reduced and controllable acoustic output, suitable for meeting modern environmental regulations. The source consists of a submersible combustion chamber filled with electrolytically generated hydrogen and oxygen ignited via spark. Upon ignition, the combustive mixture is converted into high temperature water vapor and the ensuing bubble activity radiates broadband acoustic energy. CSS has previously been used in the water column from stationary platforms, and has been deployed on the bottom to generate seismic interface waves. We now report the successful implementation of a self-contained CSS system deployed in a tow body. No hazardous gas is ever stored on board the ship, as it is produced *in situ* while at depth. The system can produce high amplitude acoustic pulses while being stably towed behind a ship with an electromechanical cable. Discussion will focus on the functionality, capability, and expandability of the system. [Work supported by ONR.]

10:20

3aUWb2. A single-crystal acoustic hydrophone for increased sensitivity. Jeremy Brown (Biomedical Eng., Dalhousie Univ., 1276 South Park St., Dickson Bldg. rm 3191, Halifax, NS B3H2Y9, Canada, j.brown@dal.ca), Kevin Dunphy, and Olivier Beslin (Res. and Development, Ultra Electron. Maritime Systems, Dartmouth, NS, Canada)

This study describes the development of a underwater surveillance hydrophone based on next generation $PbMg_{1/3}Nb_{2/3}O_3$-$PbTiO_3$ (PMN-PT) single-crystal piezoelectric as the hydrophone substrate. Although PMN-PT can possess much higher piezoelectric sensitivity than traditional PZT piezoelectrics, it is highly anisotropic and therefore there is a large gain in sensitivity only when the crystal structure is oriented in a specific direction. Because of this, simply replacing the PZT substrate with a PMN-PT cylinder is not an optimal solution because the crystal orientation does not uniformly align with the circumferential axis of the hydrophone. Therefore, we have developed a novel composite hydrophone that maintains the optimal crystal axis around the hydrophone circumference. An 11.3 mm diameter composite hydrophone cylinder was fabricated from a single <110> cut PMN-PT rectangular plate. Solid end caps were applied to the cylinder and the sensitivity was directly compared with a solid PZT-5A cylindrical hydrophone of equal dimensions in a hydrophone test tank. The charge sensitivity showed a 7.5 dB improvement over the PZT hydrophone and the voltage sensitivity showed a 2.7 dB improvement. This was in relatively

good agreement with the theoretical improvements of 12.88 dB and 5.44 dB, respectively.

10:40

3aUWb3. The shallow sea experiment with usage of linear hydrophone array. Wojciech Szymczak (Hydroacoustic Inst., Polish Naval Acad., str. Smidowicza 69, Gdynia 81-103, Poland, ws2@o2.pl), Eugeniusz Kozaczka (Gdansk Univ. of Technol., Gdynia, Poland), Grazyna Grelowska, Ignacy Gloza, and Sławomir Kozaczka (Hydroacoustic Inst., Polish Naval Acad., Gdynia, Poland)

Purpose of this article is to present designed and made linear hydrophone array and the results obtained during *in situ* trails on Gulf of Gdańsk. The measuring system allowed to localize hydrophones in the selected points and perform measurements in both the horizontal antenna positioning and vertical. Made in this way recordings allow creating accurate 3D imaging of sound intensity/propagation. During research three floating objects were measured: small ship (18 m long), yacht (12 m long) and 5 m pontoon with engine and paddles used to drive. In the article, accurately will be described the entire measurement system and complementary devices (navigation system, sound speed profiler, online underwater monitoring to control linear antenna position) and procedures used during *in situ* measurement circuit check and calibration using Lubell Underwater Speaker with amplifier and connected generator with set of reference signals. Characteristic arrangement of sensors allows use of hyperbolic navigation algorithms which results will be presented with an emphasis on measurements when the unit performed circulation around the measurement system. Furthermore, some spectrograms, cross correlations and frequency classification dependence of the speed using a prepared script in MATLAB programing environment will be discussed and presented.

11:00

3aUWb4. Laboratory investigation with subbottom parametric echosounder SES-2000 standard with an emphasis on reflected pure signals analysis. Wojciech Szymczak, Grazyna Grelowska (Hydroacoustic Inst., Polish Naval Acad., str. Smidowicza 69, Gdynia 81-103, Poland, ws2@o2.pl), Eugeniusz Kozaczka (Gdansk Univ. of Technol., Gdynia, Poland), and Sławomir Kozaczka (Hydroacoustic Inst., Polish Naval Acad., Gdynia, Poland)

The main goal of the paper is to describe correlations between measurements results of trials taken on Guls of Gdańsk bottom sounded with parametric echosounder SES-2000 Standard and laboratory research where collected during survey sediments were measured. Stationary tests took place at Gdańsk University of Technology where 30 meters long 1.8 meter deep and 3 meters wide water tank is located. Main lobe of antenna was directed parallel to the longest dimension. Hydrophones used during experiment were fixed to the 3D positioning system—ISEL, which gave the opportunity to place sensor with high precision in the middle of main lobe or other specified places. Using prepared to this experiment frames different sea bottom layers configurations corresponding to the natural structure were sounded. Data obtained during laboratory measurements and trials *in situ* were combined to draw conclusions about proper interpretation of echograms and begin the process of sediments classification. Analyses were done with MATLAB programing software were data were imported and used to the simulations and comparisons.

11:20

3aUWb5. Environmental acoustic parameters of the Sea of Japan shelf (Peter the Great Gulf). Alexandr N. Samchenko, Igor O. Yaroshchuk, and Alexandra V. Kosheleva (V.I.Il`ichev Pacific Oceanological Inst., 43, Baltiyskaya Str, Vladivostok, 690041, Russia, samchenko.alexandr@yandex.ru)

The paper describes general geoacoustic model of the Peter the Great Gulf (the largest gulf of the Sea of Japan) and detailed model of its part of 400 square km in size. The general geoacoustic model is formed on the basis of geological, seismic, and bathymetric data. It includes distribution of P-, S-wave, density, and attenuation of friable sediments at the bottom surface and averaged characteristics for different types of rocks in the gulf. Bathymetric data processing was carried out by means of two-dimensional singular spectrum analysis, based on the ratio between the size of relief structures and the energy expended under the development of tectonic processes. Detailed geoacoustic model is based on processing of hydroacoustic and seismic authors' studies (3.5 kHz sonar, 50–100 kHz sonar and air-gun). Three sedimentary layers and the location of upper edge of the granite stratum are marked in the observable shelf area. The model demonstrates significant changes in P-, S-wave in the sediments of the lateral (1500–1750 m/s—P-wave and 120–600 m/s—S-wave) and vertical (1500–5400 m/s—P-wave and 120–3300 m/s—S-wave). Some of the most important oceanographic mechanisms determining both large- and small-scale spatio-temporal variations of sound speed field in the same area are presented. These data were obtained by the authors within last several years. Qualitative analysis of the propagation of low-frequency signals is presented as an example of application of the model.

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 513ABC, 1:00 P.M. TO 2:00 P.M.

Session 3pAAa

Architectural Acoustics and Musical Acoustics: Virtual Concert Hall Acoustics II

Sungyoung Kim, Cochair
RIT, ENT-2151, 78 Lomb Memorial Dr., Rochester, NY 14623

Wieslaw Woszczyk, Cochair
Music Res., McGill Univ., Schulich School of Music, 527 Sherbrooke St. West, Montreal, QC H3A1E3, Canada

Contributed Papers

1:00

3pAAa1. Is there really a whispering gallery at the Great Ballcourt at Chichen Itza, Mexico? David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683-4514, dlubman@dlacoustics.com)

A "whispering gallery" (WG) at the Great Ballcourt (GBC) was first reported during its excavation in the 1920s by American archaeologist Silvanus Morley (1883–1948), Director of the Carnegie Institution's Chichen Itza project. In his 1925 National Geographic article Morley wrote: "Standing in this temple one can speak in a low voice & be heard distinctly at the other end of the court, 500 ft away." Around 2000–2001, queries on AZTLAN, a semi-official Mesoamerican archaeology Internet discussion group, found little or no belief in a WG by mesoamericanists. Some opined that any WG would surely be a design accident or an artifact of ballcourt ageing or reconstruction. They stiffened at the suggestion that the ancient Maya might have possessed the requisite knowledge for intentional design.

Was Morley mistaken? Or are modern mesoamericanists missing something? During a tour of Chichen Itza following the fall 2002 joint acoustical meeting in Cancun, Mexico, the author and two of his colleagues convincingly demonstrated a GBC WG to about 100 acousticians and their companions. This paper describes WG phenomena observed at the Great Ballcourt and suggests physical models to explain them. He also presents evidence for intentional design.

1:20

3pAAa2. Three-dimensional sound spatialization at Auditorio400 in Madrid designed by Jean Nouvel. Emiliano del Cerro and Silvia M[a] Ortiz (TIC, Universidad Alfonso X el Sabio, Avenida Universidad 1, Madrid 28691, Spain, ecerresc@uax.es)

The auditorium 400 was designed by the team of Jean Nouvel, the French architect, Pritzker Prize winner in 2008. It belongs to the organization of the National Museum and Reina Sofia Art Centre in Madrid, and is incorporated in a special room within the cultural life of Madrid (Spain). The center collaborates with the Spanish Ministry of Culture, and organizes a series of concerts of contemporary music and electronic and computer music. To achieve the sound projection equipment, the direction of the audience chose the system Acousmonium, designed by the GRM inside the ORTF in Paris. This paper will explain the involvement of the group LIEM (laboratory for Computer and Electronic Music), from Reina Sofia Museum with space in music: The musical relationships and implications of this choice, as well as the technical, architectural, and signal processing techniques used for the design of algorithms for spacialization of sound. After giving a very general overview of specific algorithms for spacialization, we explain some musicals examples designed specifically for this space, and the impact of its implementation and diffusion in the auditorium400 that have very special technical and artistic features.

1:40

3pAAa3. Effect of acoustic and visual stimuli on preference for different seating positions in a concert hall and an opera theater. Shin-ichi Sato, Adrián Saavedra, Alejandro Bidondo (Ingeniería de Sonido, Universidad Nacional de Tres de Febrero, Varentín Gómez 4752, Caseros, Buenos Aires 1678, Argentina, ssato@untref.edu.ar), Shuo Wang, Yuezhe Zhao, Shuoxian Wu (State Key Lab. of Subtropical Bldg. Sci., South China Univ. of Technol., Guangzhou, China), Nicola Prodi, and Roberto Pompoli (Dipartimento di Ingegneria, Università degli Studi di Ferrara, Ferrara, Italy)

The sound fields and the views of several positions in a concert hall and an opera theater were simulated and the subjective preference for different seating positions was investigated. First, the seat preference with and without visual stimuli under the conditions of (1) the original sound level (the sound pressure level at each position was maintained as the impulse response measurements in the auditoria) and (2) the equalized sound level were compared since the subjective scale of seat preference showed the highest correlation with sound level in the previous study investigating the opera theater [Sato *et al*., Acustica united with Acta Acustica **98**, 749–759 (2012)]. Some positions were judged acoustically preferred but visually less preferred or vice versa. Thus, another preference test was conducted by using the combinations of the acoustic and the visual stimuli of different positions to further investigate the audio-visual interaction.

Demonstration

The organizers of "Virtual Concert Hall Acoustics" have arranged a demonstration and a live music concert using virtual acoustics technology of McGill University. The demo and concert will take place Wednesday evening in the Multimedia Room (MMR) at the Schulich School of Music of McGill University. For information, please contact Sungyoung Kim (sxkiee@rit.edu) or Wieslaw Woszczyk (wieslaw@music.mcgill.ca).

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 513DEF, 1:00 P.M. TO 3:00 P.M.

Session 3pAAb

Architectural Acoustics: Balancing Risk and Innovation in Acoustical Consulting

Eric L. Reuter, Chair
Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801

Invited Papers

1:00

3pAAb1. Risk and innovation—Following on from the 2012 Knudsen Lecture, recent experience with calculated risk for the purpose of creating remarkable spaces is reviewed. Scott D. Pfeiffer (Threshold Acoust. LLC, 53 West Jackson Blvd., Ste. 815, Chicago, IL 60604, spfeiffer@thresholdacoustics.com)

Owner expectations, architectural vision, and creation of the intended aural environment often come together at a critical point in the design process. Frequently the answer that satisfies all of the requirements stretches the comfort level of all parties. When balanced properly, this element of risk frees the entire project team and reaches unexpected, but welcome, outcomes. The author will illustrate these critical decision points in several project examples, outlining the calculation in the calculated risk, and the opportunity to deepen the confidence in the consulting process through further research.

1:20

3pAAb2. Letters from the edge: Less conventional acoustical solutions. Jack B. Evans (JEAcoustics, Engineered Vib. Acoust. & Noise Solutions, 1705 West Koenig Ln., Austin, TX 78756, Evans@JEAcoustics.com)

The essence of acoustical analysis and problem solving still is source–path–receiver; yet, as standards, criteria and products evolve, solution precedents should be revalidated or invalidated, so that new, innovative approaches may be considered. While weighing the best interests of the client, should one rigidly follow procedures developed long ago for average or anticipated conditions, or might the conventional wisdom not always be correct? This paper presents a series of short case-studies where noise or vibration were treated at the source and/or along the path, but with some "twist" or variation, either from typical solution applications or owner/client response. Relevant standards, ordinances, or criteria are referenced. Where available, on-site acoustical measurements, observations, photos, or receiver experiences illustrate concepts. For the case studies presented: indicate (i) importance of the problem, (ii) method of the development used for problem solving, (iii) original contribution of the work, and (iv) conclusions. Case studies may include any of the following: standby power generators for data center, university cafeteria, and dining hall, coffee grinding and packaging shop within a grocery market, exercise/therapy floor impact above offices, air-cooled refrigeration chillers near residences, high-rise elevator equipment room adjacent to residential units, and/or high-rise domestic water booster/circulation pump room adjacent to residential unit.

1:40

3pAAb3. Design and implementation of a tuned low-frequency absorber in a residential music listening/practice room. Aaron M. Farbo and Christopher A. Storch (Cavanaugh Tocci Assoc., 327F Boston Post Rd., Subury, MA 01776, afarbo@cavtocci.com)

The historic Bradley Mansion in Boston's Back Bay neighborhood was recently completely renovated and subdivided into multiple luxury-level condominium units built to suit to the new homeowners requirements. The historic preservation requirements imposed on the project presented challenges for designing and implementing sufficient acoustic absorption and sound isolation for the design of a new music listening and piano practice room in the lowest level residence. A custom-designed low-frequency absorber was developed to fit within an existing architectural niche at one end of the elliptical music room, and additional absorptive treatment was added to the walls to blend in with the existing finishes. Sound isolation to the neighboring condominium above the music room was improved via floor/ceiling modifications—a task made more challenging by the need to retain the terracotta structure and other historic details. Construction details and measurement results will be discussed in this case study presentation.

2:00

3pAAb4. A statistical analysis of acoustical measurement uncertainties for assemblies within multi-family dwellings in the United States. David W. Dong and John LoVerde (Veneklasen Assoc., 1711 16th St, Santa Monica, CA 90404, wdong@veneklasen.com)

Over the last several years, the authors have demonstrated that the uncertainties in the methods of acoustical testing are much larger than realized by professionals and lay people [LoVerde and Dong, J. Acoust. Soc. Am. **125**, 2629 (2009); J. Acoust. Soc. Am. **126**, 2171 (2009); J. Acoust. Soc. Am. **130**, 2355 (2011)]. Acceptance of the large uncertainty immediately raises many practical questions. How much of the uncertainty is inherent in the test procedure and how much is due to differences between laboratories, installation methods, contractors, materials, etc.? Is the data hopelessly chaotic, or is there "true value" that can be obtained by suitable data processing? How many tests are required to feel confident in the characterization of an assembly? Some rules of thumb have been developed based on our experience [LoVerde and Dong, J. Acoust. Soc. Am. **131**, 3319 (2012)], but these questions have not yet been systematically addressed before. A statistical analysis has been performed using a database of thousands of laboratory and field noise isolation tests. Results are presented that address these questions.

2:20

3pAAb5. Coping with curves in room design. Timothy Foulkes (Cavanaugh Tocci Assoc., 327 Boston Post Rd., Sudbury, MA 01776, tfoulkes@cavtocci.com)

Concave curves are pleasing to the eye, but they cause a number of different acoustic anomalies. Depending on the radius of curvature, finish material, included angle, and position relative to source and receiver, one may hear a strong return echo, a noticeable coloration of the frequency balance, a dramatic shift in the acoustic image, or extended reverberation at low frequencies. The effects of standard curved forms such as the Capital rotunda are well known. These anomalies are of little consequence in a transient space such as a lobby, but are problematic in rooms for speech and music presentation. Covering the entirety of the concave surface with sound absorbing material is not always the best solution. In most cases, the client will want to know the minimum acoustic treatment to avoid complaints. The author will present a series of case studies showing room designs with concave curves and the acoustic solutions.

Contributed Paper

2:40

3pAAb6. Building information modeling and the consultant: Managing roles and risk in an evolving design and construction process. Norman H. Philipp (Geiler & Assoc., 1840 E. 153rd Circle, Olathe, KS 66062, nphilipp@geileracoustics.com)

Paramount changes are occurring within the building and construction industries, fueled by the ever expanding abilities of computer modeling technologies. This revolution not only impacts our approach to and execution of the physical design of a building, but also the construction and the day to day management of the facilities. Current technologies have allowed the Building Information Modeling (BIM) process to replace many of the time tested design methods of the past. With this shift to new technologies also come new risks which require recognition by the acoustical consultant to ensure our evolution to meet the new paradigm of the current design environment. In this paper the importance of understanding the purpose and role of a BIM implementation/execution plan will be covered inclusive of defining the role, responsibilities, and risks associated with the acoustical consultant.

Session 3pAB

Animal Bioacoustics and Psychological and Physiological Acoustics: Perceiving Objects II

Caroline M. DeLong, Cochair
Psychology, Rochester Inst. of Tech., 18 Lomb Memorial Dr, Rochester, NY 14623

Eduardo Mercado, Cochair
Dept. of Psych., Univ. at Buffalo, Buffalo, NY 14260

Chair's Introduction—12:55

Invited Papers

1:00

3pAB1. Categories, concepts, and calls: Auditory perceptual mechanisms and cognitive abilities across different types of birds. Allison H. Hahn, Lauren M. Guillette, Marisa Hoeschele (Psychology, Univ. of Alberta, P217 Biological Sci. Bldg., Edmonton, AB T6G2E9, Canada, ahhahn@ualberta.ca), Robert G. Cook (Psychology, Tufts Univ., Medford, MA), and Christopher B. Sturdy (Psychology, Univ. of Alberta, Edmonton, AB, Canada)

Although involving different animals, preparations, and objectives, our laboratories (Sturdy's and Cook's) are mutually interested in category perception and concept formation. The Sturdy laboratory has a history of studying perceptual categories in songbirds, while Cook laboratory has a history of studying abstract concept formation in pigeons. Recently, we undertook a suite of collaborative projects to combine our investigations to examine abstract concept formation in songbirds, and perception of songbird vocalizations in pigeons. This talk will include our recent findings of songbird category perception, songbird abstract concept formation (same/different task), and early results from pigeons' processing of songbird vocalizations in a same/different task. Our findings indicate that (1) categorization in birds seems to be most heavily influenced by acoustic, rather than genetic or experiential factors (2) songbirds treat their vocalizations as perceptual categories, both at the level of the note and species/whole call, (3) chickadees, like pigeons, can perceive abstract, same-different relations, and (4) pigeons are not as good at discriminating chickadee vocalizations as songbirds (chickadees and finches). Our findings suggest that although there are commonalities in complex auditory processing among birds, there are potentially important comparative differences between songbirds and non-songbirds in their treatment of certain types of auditory objects.

1:20

3pAB2. Reverberlocation in chickadees? Eduardo Mercado, Matthew W. Wisniewski, Brittany E. McIntosh (Dept. of Psych., Univ. at Buffalo, Buffalo, NY 14260, emiii@buffalo.edu), Lauren M. Guillette, and Christopher B. Sturdy (Dept. of Psych., Univ. of Alberta, Edmonton, AB, Canada)

Chickadee songs provide conspecifics with information about the locations of singers. Song amplitude, frequency, and reverberation all vary with distance, and it is thought that chickadees use such cues to estimate distance. The current study examined transmission of chickadee songs in an open field to assess whether other cues such as relative changes in inter-note timing or relative differences in spectral energy might also provide useful information about a singer's location. Surprisingly, the difference between direct signal energy and reverberant spectral energy provided clear indications of how far a song had traveled. Preliminary analyses suggest that this cue may be robust to variations in source level, note duration, note frequency, and transmission loss. If chickadees use this cue to judge auditory distance, then this may explain why they maintain specific spectral ratios between the notes within their songs. Specifically, the spectral spacing of notes within songs appears to be directly related to chickadee auditory filter bandwidth. We describe ranging of a singing chickadee based on the spectral profile of its songs as reverberlocation (construed as an instance of passive echolocation) because it involves comparisons between a direct signal and echoes of a signal.

Contributed Papers

1:40

3pAB3. Localization of flying insects by echolocation. Ikuo Matsuo (Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, matsuo@cs.tohoku-gakuin.ac.jp) and Takuma Takanashi (Forestry and Forest Products Res. Inst., Tsukuba, Japan)

Using the echolocation, bats can capture insects in real 3D space. Bats can accurately localize these objects from echoes by emitting the frequency modulation sound. The object's range could be estimated from delay times between the emitted sound and echoes from objects. In the case of flying insects, the echoes were influenced by Doppler shift, that is, the wing beats and flight speed. In the case of the linear frequency modulated (LFM) sound, this range accuracy was dependent on not only the frequency width of emitted sound but also the Doppler shift. It has been shown that the previous proposed model could accurately estimate each range of static objects by using the frequency modulation sound. However, it was unknown whether this model could estimate locations and movements of the flying insect. In this study, the echoes were measured from the flying insect by emitting intermittently the LFM sounds. At the same time, the movements of the insects were measured by the camera. The time-frequency pattern

were computed by using the convolution of the chirplet filters. It was examined that the the insect's positions were estimated by extracting the onset from the time-frequency pattern. [Research supported by JST, CREST.]

2:00

3pAB4. Analysis of Northern bottlenose whale pulses and associated reflections recorded from the Gully Marine Protected Area. Bruce Martin (Halifax, JASCO Appl. Sci., 32 Troop Ave., Ste. 202', Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com) and Hilary Moors-Murphy (Bedford Inst. Of Oceanogr., Dept. of Fisheries and Oceans, Dartmouth, NS, Canada)

The Gully Marine Protected Area (MPA) is a large submarine canyon at the edge of the Scotian Shelf, south of Nova Scotia. A resident population of northern bottlenose whales are known to occur in the Gully throughout the year, and the canyon provides important foraging grounds for the population. Bottom-mounted Autonomous Multichannel Acoustic Recorders (AMAR) were deployed in the Gully for ten days in in March 2010 (sampling rate = 375 ksps) and two days in October 2011 (sampling rate = 128 ksps). Bisonar pulses produced northern bottlenose whales (likely used to echolocate prey) were recorded consistently throughout these AMAR deployments. The swept FM characteristics of the northern bottlenose whale pulses recorded were consistent over both years, and both data sets contained clear pulse reflections from bottom clutter or prey targets. In this paper, we provide a description of the northern bottlenose whale pulses recorded in the Gully and make recommendations on short-time Fourier transform parameters for analysis of the pulses. A description of the pulse reflections is also provided, based on analysis of the reflection patterns using short-time Fourier transforms and by matched filtering with the direct arrival from the whales.

2:20

3pAB5. Preliminary results from collaborative referring to impulsive sonar sounds. Charles F. Gaumond (Acoust. Div., Naval Res. Lab., Code 7162, 4555 Overlook Ave. SW, Washington, DC 20375, charles.gaumond@nrl.navy.mil), Derek Brock, and Christina Wasylyshyn (Information Technol. Div., Naval Res. Lab., Washington, DC)

A recent experiment is described wherein pairs of listeners (a "director" and a "matcher") collaboratively refer to eight-element sets of impulsive sonar sounds, which are the same, but ordered differently for each listener. The sounds in a given set are privately displayed on each listener's computer as a line of blank cards that play a corresponding sound when clicked and can be rearranged from left to right. The listeners' task is to move the matcher's sounds into the same order as the director's. Through conversation, the listeners work out how to verbally characterize the sounds and develop a shared vocabulary. This vocabulary is presented for selected participants and is shown to generally consist of names, actions, and properties of familiar, everyday auditory events. In general, these references function as classes and descriptors. Classes correspond to causal categories that are aurally analogous to (i.e., homophonic with) the acoustic origins of the impulsive sonar sounds. Similarly, descriptors, distinguish between the properties of signal processing features that are appropriate to impulsive sounds within a given category. [Research funded by the Office of Naval Research.]

2:40

3pAB6. An auditory perception of changes in the intensity of pulses, presented in complicated sound complex. Liudmila K. Rimskaya-Korsakova (Lab. of Bioacoustics, N.N. Andreev Acoust. Inst., Shvernika. st 4, Moscow 117036, Russian Federation, lkrk@mail.ru)

The auditory system of humans and animals is able to detect and discriminate high frequency pulses in a complicated sound complex. The purpose of the work was to find new examples of a facilitation the discrimination of intensity (or level, defined by a peak amplitude) of a test pulses, presented under composite masking conditions, and to find the possible mechanisms underlying the facilitation. The discrimination tended to deteriorate if the test pulse was presented through 50 ms after a pulse's masker. However, if the test pulse was mixed with stationary noise, the beginning of which coincided with the end of the pulse's masker, discrimination became better. The noise levels, at which facilitation occurred, depended on amplitudes of both the test pulses and the pulse's maskers. When the duration of the noise was less than 50 ms, an auditory adaptation could not influence on the discrimination. The reason of the facilitation could be in the temporal redistribution of the auditory nerve fibers activities, which occurred at coding of the complicated sound complex "pulse's masker - test pulse - stationary noise."

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 510D, 1:20 P.M. TO 2:40 P.M.

Session 3pAO

Acoustical Oceanography: Ocean Acoustical Tomography

Lora J. Van Uffelen, Chair
Univ. of Hawaii at Manoa, 1000 Pope Rd., MSB 205, Honolulu, HI 96815

Contributed Papers

1:20

3pAO1. Influence on sound spread considering the flow velocity in a horizontal layer media. Yang Song and Zhenqi Zhao (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin 150001, China, syang@163.com)

The marine currents could often influence sound propagation underwater. The traveling time of sound ray is distinctly effected by the velocity of current in some conditions. So it will be made certain errors in seeking eigen rays and inversing sound speed profile if the velocity of the current is ignored. In order to improve the computation accuracy of sound ray model, a sound ray model of horizontal layer is induced in which the media flow is considered. Eigen rays are searched and their traveling time is calculated by the ray model. It is also discussed that the velocity of flow media influences ray trace and traveling time. An average sound speed profile measured under a shallow water is cited to calculate the eigen rays. The differences of sound ray are given under two kinds of condition which the velocity of current is considered and not considered. The computation results show that the sound ray trace is changed indistinctively under small Mach number condition, but the traveling time of eigen rays is fluctuated obviously. The fluctuation of ray traveling time is bigger according to the larger Mach number and the longer spread distance. The results of study will provide some help in the inversion of ocean acoustic tomography.

1:40

3pAO2. Time-angle ocean acoustic tomography using sensitivity kernels: The forward problem. Florian Aulanier, Barbara Nicolas (Gipsa lab, Institut Polytechnique de Grenoble, 11 rue des Mathématiques, Grenoble Campus BP46 F, SAINT MARTIN D'HERES Cedex 38402, France, Florian.Aulanier@gipsa-lab.grenoble-inp.fr), Philippe Roux (ISTerre, Observatoire des Sci. de l'Univers, Université de Grenoble, Grenoble, France), and Jérôme I. Mars (Gipsa Lab., Institut Polytechnique de Grenoble, Grenoble, France)

Broadband acoustic signals around 1 kHz propagate through shallow water oceanic waveguides of ~100 m in depth and ~2 km in range as multiple ray-like wavefronts. These acoustic arrivals can be characterized by the following observables: travel-time (TT), direction-of-arrival (DOA), and direction-of-departure (DOD). By applying double-beamforming on the point-to-point signals recorded between two source-receiver arrays, the acoustic contribution of each arrival can be separated from the multi-reverberated data and the TT, DOA, and DOD observable variations are accurately measured. This study deals with the use of time-angle sensitivity kernels (TASK) to estimate the observable variations induced by sound speed perturbations in the waveguide. This approach is based on the first order Born approximation and takes into account the finite-frequency effects associated with wave propagation. The robustness the TASK approach is analyzed and compared to numerical parabolic equation simulations involving different sound speed perturbations. For example, parameters such as the perturbation location, the value and shape of the perturbation in the waveguide are modified. The combination of several perturbations and the influence of the source-receiver array apertures on the TT, DOA, and DOD estimates are also studied.

2:00

3pAO3. Time-angle ocean acoustic tomography using sensitivity kernels: Numerical and experimental inversion results. Florian Aulanier, Barbara Nicolas (Gipsa Lab, Institut Polytechnique de Grenoble, 11 rue des Mathématiques, Grenoble Campus BP46 F, SAINT MARTIN D'HERES Cedex 38402, France, Florian.Aulanier@gipsa-lab.grenoble-inp.fr), Philippe Roux (ISTerre, Observatoire des Sci. de l'Univers, Université de Grenoble, Grenoble, France), Romain Brossier (ISTerre, Observatoire des Sci. de l'Univers, Université de Grenoble, Saint Martin D'Heres, France), and Jérôme I. Mars (Gipsa Lab, Institut Polytechnique de Grenoble, Grenoble, France)

In shallow water acoustic tomography, broadband mid-frequency acoustic waves (1 to 5 kHz) follow multiple ray-like paths to travel through the ocean. Travel-time (TT) variations associated to these raypaths are classically used to estimate sound speed perturbations of the water column using the ray theory. In this shallow water environment, source and receiver arrays, combined with adapted array processing, provide the measurement of directions-of-arrival (DOA) and directions-of-departure (DOD) of each acoustic path as new additional observables to perform ocean acoustic tomography. To this aim, the double-beamforming technique is used to extract the TT, DOA, and DOD variations from the array-to-array acoustic records. Besides, based on the first order Born approximation, we introduce the time-angle sensitivity kernels to link sound speed perturbations to the three observable variations. This forward problem is then inversed by the maximum *a posteriori* method using both the extracted-observable variations and the proposed sensitivity kernels. Inversion results obtained on numerical data, simulated with a parabolic equation code, are presented. The inversion algorithm is performed with the three observables separately, namely TT, DOA, and DOD. The three observables are then used jointly in the inversion process. The results are discussed in the context on ocean acoustic tomography.

2:20

3pAO4. Toward subsurface positioning of gliders using fixed acoustic tomography sources. Lora J. Van Uffelen, Eva-Marie Nosal, Bruce M. Howe, Glenn S. Carter (School of Ocean and Earth Sci. and Technol., Univ. of Hawaii at Manoa, 1000 Pope Rd., MSB 205, Honolulu, HI 96815, loravu@hawaii.edu), Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Kevin D. Heaney, Richard L. Campbell (OASIS, Inc., Fairfax, VA), and Patrick S. Cross (OASIS, Inc., Honolulu, Hawaii)

Acoustic Seagliders can be positioned precisely using GPS at the surface, but are underwater and unable to utilize GPS for up to 9 h at a time as they dive to depths of up to 1000 m. During this time, a kinematic model estimates the position of the glider. Four acoustic Seagliders were deployed in the Philippine Sea November 2010–April 2011, and received transmissions from five broadband acoustic tomography sources moored in the region. Over 2000 acoustic receptions were recorded at ranges up to 700 km from the moored sources. Measured acoustic arrival peaks were unambiguously associated with ray arrivals predicted using the model-estimated glider position at the time of reception and a mean sound-speed profile. Estimates of source-receiver range uncertainty were calculated from statistics of travel-time offsets between the measured arrivals and the eigenray dispersion patterns. The uncertainty in range between the source and the modeled glider position during a dive is estimated to be 639 m (426 ms) rms disregarding the effects of ocean sound-speed variability, which are anticipated to be on the order of 70 ms rms.

Session 3pBA

Biomedical Acoustics: Biomedical Acoustics Best Student Paper Award Poster Session

Kevin J. Haworth, Chair
Univ. of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with USD$500 for first prize, USD$300 for second prize, and USD$200 for third prize. The award winners will be announced at the meeting of the Biomedical Acoustics Technical Committee. Below is a list of students competing, with their abstract numbers and titles listed. Full abstracts can be found in the oral session associated with the abstract numbers.

All entries will be on display and all authors will be at their posters from 1:00 p.m. to 3:00 p.m.

1aBA3. A contrast source inversion method for breast cancer detection. Student author: N. Ozmen-Eryilmaz

1aBA7. Electromagnetic hydrophone for high-intensity focused ultrasound measurement. Student author: Pol Grasland-Mongrain

1pBAa4. Sound speed estimation in single cells using the ultrasound backscatter power spectrum. Student author: Eric M. Strohm

1pBAa6. An analysis of the acoustic properties of the cell cycle and apoptosis in MCF-7 cells. Student author: Maurice M. Pasternak

1pBAb2. Acoustic and optical characterization of targeted ultrasound contrast agents. Student author: Camilo Perez

1pBAb4. Radiation for bubble contrast agents in inhomogeneous media. Student author: Chrisna Nguon

1pBAb5. Temporal evolution of subharmonic emissions from a lipid-encapsulated contrast agent. Student author: Himanshu Shekhar

1pBAb6. Simulations of transcranial passive acoustic mapping with hemispherical sparse arrays using computed tomography-based aberration corrections. Student author: Ryan Jones

1pBAb11. Passive acoustic mapping of magnetic microbubbles in an *in vitro* flow model. Student author: Calum Crake

1pBAb12. A two-component speckle model for detection of microbubble signals in linear contrast-enhanced ultrasonography. Student author: Matthew R. Lowerison

2aBA9. Ultrasonic atomization: A mechanism of tissue fractionation. Student author: Julianna C. Simon

2pBAa7. Investigating the sensitivity of microbubble acoustic response for biosensing applications. Student author: Caroline J. Harfield

2pBAa8. Modeling of microbubbles pushed through clots via acoustic radiation force. Student author: Ascanio Guarini

2pBAa11. Effect of shell thickness on sound propagation through encapsulated bubbles: A resonator approach. Student author: Craig N. Dolder

2pBAb5. Validation of three-dimensional strain tracking by volumetric ultrasound image correlation in a pubovisceral muscle model. Student author: Anna S. Nagle

2pBAb6. Measurement of surface acoustic wave in soft material using swept-source optical coherence tomography. Student author: Yukako Kato

3aBAa5. Small interfering ribonucleic acid delivery with phase-shift nanoemulsions. Student author: Mark T. Burgess

3aBAb4. Improving the acousto-optic detection of high-intensity focused ultrasound lesions. Student author: Matthew T. Adams

3aBAb6. The origins of nonlinear enhancement in *ex vivo* tissue during high intensity focused ultrasound ablation. Student author: Edward Jackson

4aBA5. Investigation on the inertial cavitation threshold of micro-bubbles. Student author: Xiasheng Guo

5aBAa3. Ultrasonic assessment of the *in vitro* biomechanical stability of a dental implant. Student author: Romain Vayron

5aBAa6. Developement and validation of resonant ultrasound spectroscopy for the measurement of cortical bone elasticity on small cylindrical samples. Student author: Simon Bernard

5aBAa8. Computational simulations of time of flight and attenuation of first arriving signal from healing process of diaphyseal femur fractures. Student author: Paulo Tadeu Rosa

5aBAb1. The effect of boundary proximity on the fundamental and subharmonic emissions from individual microbubbles at higher frequencies. Student author: Brandon Helfield

5aBAb2. Bifurcation structure of the ultrasonically excited microbubbles undergoing buckling and rupture. Student author: Amin Jafari Sojahrood

5aBAb5. Temporal and spatial characteristics of nonlinear acoustic field generated by an extracorporeal shockwave therapy device: Modeling and measurements. Student author: Maria Karzova

WEDNESDAY AFTERNOON, 5 JUNE 2013 512AE, 12:55 P.M. TO 3:00 P.M.

Session 3pEA

Engineering Acoustics: Computational Methods in Transducer Design, Modeling, Simulation, and Optimization III

Daniel M. Warren, Chair
Knowles Electronics, 1151 Maplewood Dr, Itasca, IL 60134

Chair's Introduction—12:55

Contributed Papers

1:00

3pEA1. Coupling elastic-poroelastic material in structure-borne sound modeling. Katherina Rurkowska and Sabine Langer (Institut für Angewandte Mechanik, Technische Universität Braunschweig, Spielmannstraße 11, Braunschweig, Niedersachsen 38106, Germany, infaminfo@tu-braunschweig.de)

Porous materials are widely used in noise reduction applications. To minimize the external noise produced by aircraft propeller drives, porous materials are implemented. As a part of the project *Sonderforschungsbereich 880* "Fundamentals of High Lift for Future Civil Aircraft," porous surfaces are used in the High-lift configuration to mitigate the flow noise and to influence the structure-borne sound. In order to model the performance of the applied poroelastic material, an approach coupling a poroelastic material with an elastic structure using Finite Element Method is presented. The Biot's theory is used to model the poroelastic material. The aim of this work is to simulate the effect of the entry and transmission of the structure-borne sound into the poroelastic medium. An example of the implemented model shows the plausibility of presented approach.

1:20

3pEA2. Numerical investigation of the functionally graded materials by the interaction of the plate guided waves with discontinuities and cracks. Farouk Benmeddour, Emmanuel Moulin, and Jamal Assaad (OAE Dept., IEMN, CNRS UMR 8520, Univ. of Valenciennes and Hainaut Cambrésis, Campus Mont Houy, Valenciennes 59313, France, farouk.benmeddour@univ-valenciennes.fr)

This work intends to give a better comprehension of the guided wave interactions with damage in a functionally graded material (FGM). The propagation and interactions of plate guided waves with discontinuities in a FGM composed of ceramic and metal mixture are investigated. For this purpose, a two dimensional finite element (FE) method is used to analyze the near field surrounding the damage. Then, expansion of the solution into sums of guided modes enables the determination of the reflection and transmission coefficients of each existent mode. The determination of the modal features is ensured by the way of the so called semi-analytical finite element (SAFE) method applied to the one dimensional inlet and outlet cross-sections. The latter has the benefit to study an arbitrary shape-like damage in an infinite structure having the same shape by translation in the propagation direction in a fast and efficient way. Results are obtained by solving the global system of the 2D hybrid FE-SAFE method. Different symmetrical and asymmetrical notches are studied and so for cracks. Results are achieved and discussed for a FGM and compared to those obtained for an isotropic material.

1:40

3pEA3. Generalized Debye series expansion to improve the non-destructive testing and health monitoring of cylindrical structures by guided waves. Slah Yaacoubi (Institut de Soudure, Yutz, France), Marc Deschamps, Eric Ducasse (I2M, Bordeaux, France), Laurent Laguerre (IFSTTAR, Bouguenais, France), Weina Ke Yaacoubi (Institut de Soudure, Yutz, France), Peter McKeon (Georgia Insitute of Technol., GTL, Metz, France, peter.mckeon@gatech.edu), Salah Ramadan (Institut de Soudure, Yutz, France), and Nico F. Declercq (Georgia Insitute of Technol., Metz, France)

Many structures in civil engineering notably bridges and nuclear power plants must be regularly, strictly, and carefully tested to avoid any human or environmental catastrophe. Among the NDT techniques, which can be

applied, ultrasonic guided waves are a good candidate to monitor bars and cables. However, its multimodal and dispersive behaviors can limit its performances. Theoretical modeling is sometimes needed to understand the behavior of the traveling waves to improve the testing/monitoring and made a right *in-situ* decision. The aim of this paper is to derive the space-time velocity field in a cylindrical waveguide perfectly embedded in an infinite solid matrix and generated by an inside bounded beam. This beam is generated by an off-axis source. Vector Hankel transform and Fourier series are combined to decompose the inside field into infinity of elementary cylindrical waves propagating in radial direction and planar waves propagating in axial direction. Global resolution method and Generalized Debye series expansion are used both to calculate the 3D global cylindrical reflection/transmission coefficients. The method is demonstrated through a steel bar embedded in cement matrix. Simulated frequency–wavenumber diagrams show that the embedding material acts like filter for different frequency ranges. Other results will be presented.

2:00

3pEA4. The effect of a middle layer on ultrasonic wave propagating in a three-layer structure. Raymond B. Mabuza and Ngeletshedzo Netshidavhini (NDT and Phys., Vaal Univ. of Technol., Private Bag X021, Vanderbijlpark, Gauteng 1900, South Africa, raymondm@vut.ac.za)

In this paper, the focus of attention is on the effect of an elastic middle layer on the propagation behavior of ultrasonic waves. Systematic parametric studies are conducted to quantify the effects of the middle layer upon the ultrasonic wave propagation, including its thickness and acoustic impedance. We treat this problem analytically and numerically. The three-layer structure is also used to investigate the influence of the imperfect interfaces between two outer layers and a middle layer on the ultrasonic wave propagation. The theoretical analysis considers successive reflections of waves radiated by the transmitting transducer. The output signal is a superposition of successive reflections. Our results demonstrate clearly that there is significant influence of the middle layer in our three-layer problem. Various aspects of our approach are discussed and numerical examples are used to illustrate the suitability of our approach. Some details about the numerical methods employed are also given. The results are presented and discussed.

2:20

3pEA5. Comparison of finite element models simulating the interaction of ultrasonic guided waves with sites of disbonding in composites. Peter McKeon (Mech. Eng., Georgia Insitute of Technol., 2 rue Marconi, Metz 57070, France, peter.mckeon@gatech.edu), Slah Yaacoubi (Institut de Soudure, Yutz, France), Nico F. Declercq (Mech. Eng., Georgia Insitute of Technol., Metz, France), and Salah Ramadan (Institut de Soudure, Yutz, France)

Disbonding in composite structures is a serious defect which can dramatically reduce the structures' life, and can lead to catastrophic failure. To avoid this, non-destructive testing or structural health monitoring techniques are needed. One such technique involves ultrasonic guided waves, which has recently found use in this field thanks to its ability to inspect in non-accessible areas and over long distances. Numerical models are often used because they can help explain experimental results, and offer the ability to simulate different damage scenarios, predicting results with less cost than experiments. In this work, sites of disbonding between an orthotropic composite plate and an isotropic polyamide plate were modeled via the finite element method. Several methods for modeling the damage site are employed, and results are compared with experimental ones. Model types range from the introduction of a geometrical void at the interface boundary to addressing boundary conditions between adjacent layers. The amount of mode conversion after interaction with the damage site is used to evaluate the validity of each model type. Results are discussed in terms of computational effort and accuracy in predicting true physical behavior.

2:40

3pEA6. Energy flux streamlines versus acoustic rays for modeling interaction with rigid boundaries: near field of sound from a circular loudspeaker. Cleon E. Dean (Physics, Georgia Southern Univ., P.O. Box 8031, Math/Phys. Bldg., Statesboro, GA 30461-8031, cdean@georgiasouthern.edu) and James P. Braselton (Mathematical Sci., Georgia Southern Univ., Statesboro, GA)

Sound emitted by a circular loudspeaker can be treated as equivalent to a plane wave diffracted by a circular aperture in a rigid, sound absorbing screen. Axial symmetry leads one to expect constructive interference along the symmetry axis in the near field (the Poisson-Arago spot). An energy flux streamline model was developed to help visualize this and other features of the near sound field. The model is used to draw out similarities and differences between energy flux streamlines and acoustic rays, particularly in the transition to the far field.

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 510C, 1:20 P.M. TO 2:20 P.M.

Session 3pED

Education in Acoustics: Take 5's

Andrew Morrison, Chair
Natural Sci. Dept. Joliet Junior College, 1215 Houbolt Rd, Joliet, IL 60431

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

Session 3pMU

Musical Acoustics and Psychological and Physiological Acoustics: Perception and Orchestration Practice

Stephen McAdams, Cochair
Music Res., McGill Univ., 555 Sherbrooke St. W., Montreal, QC H2W 1S2, Canada

Punita G. Singh, Cochair
Sound Sense, 16 Gauri Apartments, 3 Rajesh Pilot Ln., New Delhi 110011, India

Invited Papers

1:00

3pMU1. Timbre as a structuring force in music. Stephen McAdams (Schulich School of Music, McGill Univ., 555 Sherbrooke St. W., Montreal, QC H2W 1S2, Canada, smc@music.mcgill.ca)

Most of the music we enjoy uses the musical qualities of different instruments to create specific perceptual and emotional effects that composers sculpt over time. Timbre is the auditory attribute that distinguishes different instruments. Research on timbre perception has demonstrated that it is multifaceted and contributes in many ways to the perceptual organization of musical structures. The art of structuring music with timbre is orchestration. A survey of orchestration treatises reveals the dearth of underlying theory, in sharp contrast to other traditional areas such as harmony and counterpoint, which have long theoretical traditions. We seek to develop a theoretical ground for orchestration practice starting with the structuring role that timbre can play in music. Many aspects of musical structuring are achieved by auditory scene analysis, the perceptual processes that result in unified events, integrated streams of events, groups of events segmented into phrases and sections, and larger-scale units extended over time that we call orchestral gestures. The roles that timbre plays in the manifestation of these principles in orchestration practice will be considered as potential elements of a theory of orchestration. How such principles might be incorporated into computer-aided orchestration systems and computer-aided orchestral rendering systems will also be examined.

1:20

3pMU2. Acoustic and musical features of emotional response to orchestral gestures. Meghan Goodchild (CIRMMT and McGill Univ., 4515 rue Drolet, Unit 6, Montreal, QC H2T 2G1, Canada, meghan.goodchild@mail.mcgill.ca)

Recent empirical research indicates the impact of prominent changes in instrumentation on the listening experience: several studies suggest that timbral changes evoke music-induced emotions. However, orchestration remains an underdeveloped area of music theory. A model of orchestral gestures defined by changes in instrumentation in terms of time course (gradual or sudden) and direction (addition or reduction) is presented. An exploratory behavioral study that tested the perceptual relevance of orchestral gestures on listeners' continuous ratings of emotional intensity was conducted. We demonstrate a new type of visualization that illustrates the relative textural density of each instrument family over time combined with other time-varying parameters extracted from the signal (loudness, spectral centroid, tempo, and roughness) and calculated from the score (instrumental texture, melodic range, and attack density). In addition to quantitative and qualitative comparison of similar orchestral gestures across pieces, we use this method to study the interaction of specific instrumentation changes and other musical parameters. Through discussion of the visualizations, we highlight relationships between the perceptual and musical/acoustical dimensions and quantify elements of the temporality of these experiences.

1:40

3pMU3. Perception and orchestration of melody, harmony, and rhythm on instruments with "chikari" strings. Punita G. Singh (Sound Sense, 16 Gauri Apartments, 3 Rajesh Pilot Ln., New Delhi 110011, India, punita@gmail.com)

The use of "chikari" strings on instruments such as the sitar and sarod manifests principles of Auditory Scene Analysis in creating a harmonic reference, melodic contrast, and rhythmic accompaniment. Unlike the principal "baj" strings on which the main melody is played, or resonant "tarb" strings that reinforce volume, the "chikari" strings are sounded at strategic points in performance to provide a drone, add texture, outline chords, mark rhythmic positions, and keep tempo. Listening and transcription experiments conducted with recordings of interleaved notes played on "chikari" and "baj" strings validate how differences in their timbre and tuning help to keep them perceptually apart while forming more coherent patterns based on timbre similarity and pitch proximity. Such grouping and segregation can affect the perception of temporal order, maintain the illusion of melodic continuity and in some cases of virtual polyphony. These observations add to the growing body of evidence supporting the role of timbre as a structural dimension of music and illustrate how a single instrument can bring about orchestral effects via the strategic use of devices such as "chikari" strings.

Contributed Papers

2:00

3pMU4. Predicting blend between orchestral timbres using generalized spectral-envelope descriptions. Sven-Amin Lembke (Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, sven-amin.lembke@mail.mcgill.ca), Eugene Narmour (Dept. of Music, Univ. of Pennsylvania, Philadelphia, PA), and Stephen McAdams (Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., Montreal, QC, Canada)

Composers rely on implicit knowledge of instrument timbres to achieve certain effects in orchestration. In the context of perceptual blending between orchestral timbres, holistic acoustical descriptions of instrument-specific traits can assist in the selection of suitable instrument combinations. The chosen mode of description utilizes spectral-envelope estimates that are acquired as pitch-invariant descriptions of instruments at different dynamic markings. Prominent local spectral-envelope traits, such as spectral maxima or formants, have been shown to influence timbre blending, involving frequency relationships between local spectral features, their prominence as formants, and constraints imposed by the human auditory system. We present computational approaches to predict timbre blend that are based on these factors and explain around 85% of the variance in behavioral timbre-blend data. Multiple linear regression is employed in modeling a range of behavioral data acquired in different experimental investigations. These include parametric investigations of formant frequency and magnitude relationships as well as arbitrary combinations of recorded instrument audio samples in dyads or triads. The cataloguing of generalized acoustical descriptions of instruments and associated timbre-blend predictions for various instrument combinations could serve as a valuable aid to orchestration practice in the future.

2:20

3pMU5. Timbre saliency vs. timbre dissimilarity – What is the relationship? Song Hui Chon and Stephen McAdams (CIRMMT, McGill Univ., 3484 Rue Durocher #401, Montreal, QC H2X 2E4, Canada, songhui.chon@mail.mcgill.ca)

We have proposed the notion of timbre saliency as the attention-capturing quality of timbre. The definition of saliency requires an object to stand out with respect to its surroundings, implying dissimilarity between the object and its neighbors. What then might be the relationship between timbre saliency and timbre dissimilarity? A classic timbre dissimilarity experiment and a timbre saliency experiment were carried out with 20 participants on the same set of stimuli. Multidimensional scaling revealed a two-dimensional dissimilarity space. Using the features obtained from the Timbre Toolbox [Peeters *et al.*, J. Acoust. Soc. Am. **130**, 2902–2916 (2011)], the first dimension shows a high correlation with spectral centroid [$r(13) = 0.845$, $p < 0.0001$] and spectral spread [$r(13) = 0.855$, $p < 0.0001$], both based on the ERB-FFT model spectrum, and the second with the attack time [$r(13) = -0.692$, $p = 0.004$] and power spectral crest [$r(13) = 0.732$, $p < .005$]. This confirms spectral centroid and attack time as two major acoustic correlates of timbre dissimilarity. The saliency dimension shows a moderate correlation with the second dimension [$r(13) = 0.578$, $p = 0.0241$] but not with the first dimension [$r(13) = 0.182$, $p = 0.517$], suggesting that the saliency might be more related to the temporal characteristics of timbre.

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 511BE, 1:00 P.M. TO 2:20 P.M.

Session 3pNSa

Noise, ASA Committee on Standards, Engineering Acoustics, and Structural Acoustics and Vibration: Wind Turbine Noise II

Nancy Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton St., Boston, MA 02118

Paul Schomer, Cochair
Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821

Sheryl Grace, Cochair
Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215

Contributed Papers

1:00

3pNSa1. Wind farm—Long term noise and vibration measurements. Martin Meunier (Environment, SNC-Lavalin, 2271, boul. Fernand-Lafontaine, Longueuil, QC J4G2R7, Canada, martin.meunier@snclavalin.com)

Most of the energy produced in Quebec comes from renewable sources. The concept of wind energy emerged in the late 1990's and has since become a complementary source of energy alongside hydroelectricity. Wind farms are generally seen as a good sustainable way to produce energy. However, they are not implemented without some impact on the environment. SNC-Lavalin Environment has performed many surveys in recent years for wind farm projects, including noise measurements both before and after their commissioning. This presentation will give an overview of one such project where long term noise and vibration measurements where conducted. Vibration measurements, as well as outdoor, indoor, and low frequencies noise measurements were completed both with and without the wind turbines in operation. Data will be presented showing different problems encountered in the analysis phase. For example, multiple intermittent and non-steady noise sources were present during measurement (wind turbines, car pass-bys, wind in the trees, human activities). Methods used to overcome these obstacles will be discussed (use of statistical parameters, linear regression), and the effect of the wind turbine operation on the noise level (including low frequencies) and vibration level will be presented.

1:20

3pNSa2. RoBin - A one-man measurement system for standard acoustic emission measurement according to IEC 61400-11. Daniel Vaucher de la Croix (ACOEM, 200 Chemin des Ormeaux, Limonest 69578, France, daniel.vaucherdelacroix@acoemgroup.com) and Timo Klaas (WOLFEL MESS-SYSTEME, Höchberg, Germany)

Wind turbines are built at more and more locations—which makes their noise emission an important subject. The international standard IEC 61400-11 and the German Technische Richtlinie fur Windenergieanlagen, Teil of the FGW were set up in order to unify the evaluation of noise emission. Measurement of noise emission according to these standards is linked to formidable challenges, especially for the installation of testing equipment and evaluation of data. After a short reminder on the ISO 61400 standard, the proposed paper will discuss the details of operational constraints linked with on-site measurements and how modern communication technologies help in an easy system deployment and most efficient operation for the benefits of its users.

1:40

3pNSa3. Building integrated wind turbines—A pilot study. Ben Dymock (The Acoust. Group, Dept. of Urban Eng., London South Bank Univ., 12 Deans Close, Amersham HP6 6LW, United Kingdom, dymockb@lsbu.ac.uk) and Stephen Dance (The Acoust. Group, Dept. of Urban Eng., London South Bank Univ., London, United Kingdom)

The current planning guidance in London requires that all new or refurnished large buildings should include 20% renewables. As part of a study on urban wind a pilot investigation based on the building integrated wind turbines on the skyscraper Strata Tower in London will be monitored for acoustics, vibration, anemometry and electrical generation. Strata Tower is a 150 m building in an urban location with three 19 kWe turbines in a specially designed venturi housing. The effect of the wind turbines on residents, the local community, and the building structure will be assessed and reported.

2:00

3pNSa4. Assessment of annoyance due to wind turbine noise. Malgorzata Pawlaczyk-Luszczynska, Adam Dudarewicz, Kamil Zaborowski, Malgorzata Zamojska, and Malgorzata Waszkowska (Dept. of Physical Hazards, Nofer Inst. of Occupational Medicine, 8, Sw. Teresy str., Lodz 91-348, Poland, mpawlusz@imp.lodz.pl)

The overall aim of this study was to evaluate the perception and annoyance of noise from wind turbines in populated areas of Poland. The study group comprised 378 subjects. All subjects were interviewed using a questionnaire developed to enable evaluation of their living conditions, including prevalence of annoyance due to noise from wind turbines, and the self-assessment of physical health and well-being. In addition, current mental health status of respondents was assessed using Goldberg General Health Questionnaire GHQ-12. For areas where respondents lived, A-weighted sound pressure levels (SPLs) were calculated as the sum of the contributions from the wind power plants in the specific area. It has been shown that the wind turbine noise at the calculated A-weighted SPL of 30−50 dB was perceived as annoying outdoors by about one third of respondents, while indoors by one fifth of them. The proportions of the respondents annoyed by the wind turbine noise increased with increasing A-weighted sound pressure level. Subjects' attitude to wind turbines in general and sensitivity to landscape littering was found to have significant impact on the perceived annoyance. Further studies are needed, including a larger number of respondents, before firm conclusions can be drawn.

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 511CF, 1:00 P.M. TO 2:40 P.M.

Session 3pNSb

Noise: Noise Barriers

Murray Hodgson, Chair
UBC, 2206 East Mall, Vancouver, BC V6T1Z3, Canada

Contributed Papers

1:00

3pNSb1. Effect of scaling laws for noise reduction optimization of wind fences. JohnPaul R. Abbott, Richard Raspet, and Jeremy Webster (Dept. of Phys. and Astronomy, National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., Rm. 1044, Oxford, MS 38677, johnpaul.abbott@gmail.com)

This paper will report on an investigation to increase noise reductions at low wavenumbers for a large windscreen enclosure described in two earlier papers [J. Acoust. Soc. Am. **129**, 2445 (2011); J. Acoust. Soc. Am. **132**, 2048 (2012)] by first doubling its height and then doubling its diameter. According to the scaling laws developed for small windscreens, windscreens of similar shape but differing size will have nearly identical reductions for scaled wavenumbers; therefore the wavenumbers at which noise reduction for a windscreen occurs is dependent on its size and by increasing either its height or diameter, or both, reductions should shift to lower wavenumbers. Such a shift was observed when the screen's height was doubled. Also, when scaled to height, the measured reductions for the single and double height windscreens were found to match closely, with 6 dB of reduction and greater for wave numbers between 5 and 30 1/m and max reductions of 10–13 dB.

1:20

3pNSb2. Designing canopies to improve downwind shielding at various barrier configurations at short and long distance. Timothy Van Renterghem and Dick Botteldooren (Information Technol., Ghent Univ., Sint-Pietersnieuwstraat 41, Gent 9000, Belgium, tvrenter@intec.ugent.be)

The positive effect of a row of trees to improve downwind shielding in the acoustic shadow zone behind a noise wall has been shown before by means of a wind tunnel experiment, a field study and by numerical simulations. This research focused at a rather short distance in downwind direction, where important recovery of the shielding lost by screen-induced wind refraction was observed. However, opposite effects are possible at longer distance. This can be explained by shifts in the zones with strong (positive) gradients in the horizontal component of the wind speed. Leaving a gap between the barrier top and canopy bottom helps reducing these negative effects at longer distance, and results in a generally optimized performance downwind. Trees behind noise walls at either side of the source lead to a full canceling of wind effects at short distance, but to strong negative effects at longer distances downwind. Trees as windbreaks seem to be especially useful near single, vertically erected noise walls. Near steep berms, no net effect of trees is predicted. The design rules presented in this paper are derived based on numerical calculations with a previously validated CFD-FDTD-PE model.

1:40

3pNSb3. A review of road traffic barriers for low frequency noise. Samaneh M. Fard, Nicole Kessissoglou, Stephen Samuels (School of Mech. and Manufacturing Eng., The Univ. of New South Wales, 11/127A, Barker St., Sydney, NSW 2032, Australia, fardsmb@gmail.com), and Marion Burgess (School of Eng. and Information Technol., The Univ. of New South Wales, Canberra, NSW, Australia)

Australia relies heavily on road transport due to its large area and low population density in many parts of the country. Trucks and heavy vehicles are commonly used for road freight. In addition to their normal vehicle brakes, heavy vehicles are typically fitted with release engine brakes which operate by causing the engine to act as a compressor when braking. Compression braking generates a distinct low frequency rumble that can heard at large distances and is a major source of community annoyance reactions against the heavy vehicle industry. Noise from compression brakes is an ongoing cause of complaint from many Australian residents, particularly in rural areas and at night-time. Noise barriers can be used to reduce the spread of general traffic noise and their effectiveness is determined by many factors. This paper presents a review of barriers optimized for road traffic noise and the frequency ranges at which the various barrier designs are most efficient, with a view to selecting the barriers that may be more effective at reducing the low frequency noise from compression brakes.

2:00

3pNSb4. *In-situ* measurements of sound reflection and sound insulation of noise barriers: Validation by means of signal-to-noise ratio calculations. Massimo Garai and Paolo Guidorzi (DIN, Univ. of Bologna, Viale Risorgimento 2, Bologna 40136, Italy, massimo.garai@unibo.it)

After some years from its first release, the CEN/TS 1793-5 European standard for *in-situ* measurement of sound reflection and airborne sound insulation characteristics of noise barriers has been significantly enhanced and validated in the frame of the EU funded QUIESST project. The procedure, based on impulse response measurements near the noise barrier and in the free field, is robust and easily applicable but much attention must be paid when: (i) applying the signal subtraction technique to get the reflected signal component and (ii) extracting the transmitted component, especially measuring highly insulating noise barriers. In both cases, it is essential to avoid a poor signal-to-noise ratio of the critical part of the impulse response. In the frame of the QUIESST project specific quality criteria, applicable on site, have been introduced in order to check and validate the result. These criteria are rigorously described here for the first time and illustrative examples are presented.

2:20

3pNSb5. Compliance and vegetated-barrier acoustical testing in a purpose-built sound-transmission suite. Murray Hodgson, Shira Daltrop (Acoust. and Noise Res. Group, Univ. of British Columbia, 2206 East Mall, Vancouver, BC V6T1Z3, Canada, murray.hodgson@ubc.ca), Rick Peterson, and Paul Benedict (Retaining Walls Northwest, Inc., Bellevue, WA)

Random-incidence transmission losses and absorption coefficients of a vegetated noise barrier of Criblock^{TM} construction were measured without and with plants in a sound-transmission suite built specifically for the purpose, constructed from concrete noise barriers, with the vegetated barrier separating source and receiver rooms. The suite was tested for compliance with ASTM E90-09, and found to be substantially but not completely in compliance with respect to uniformity of steady-state levels and surface absorption. It was found that the transmission loss of the vegetated barrier ranged from 42 dB at low frequencies to 66 dB at 1000 Hz; above 1000 Hz only a lower limit of the TL could be determined—values of 57–62 dB were found. These values are at least 25 dB higher than recommended by BC Ministry of Transportation guidelines. The absorption coefficients of the unplanted and planted barriers were measured; the plants decreased the absorption slightly, from NRC 0.42 to 0.37.

WEDNESDAY AFTERNOON, 5 JUNE 2013 516, 1:00 P.M. TO 3:00 P.M.

Session 3pNSc

Noise and Architectural Acoustics: Joint Poster Session on Noise and Architectural Acoustics (Poster Session)

Hideki Tachibana, Chair
Chiba Inst. of Technol., Tsudanuma 2-17-1, Narashino, Chiba 275-0016, Japan

Contributed Papers

All posters will be on display from 1:00 p.m. to 3:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:00 p.m. and contributors of even-numbered papers will be at their posters from 2:00 p.m. to 3:00 p.m.

3pNSc1. Remote keyless entry honking, convenience horn honking, and audible car alarms: Redundancies and quieter options. Jeanine Botta (Epidemiology and Biostatistics, CUNY School of Public Health at Hunter College, 1594 Metropolitan Ave., Apartment 7D, Bronx, NY 10462, jbotta@hunter.cuny.edu)

Vehicle sound emissions, car alarms, and horn honking are the subject of many noise complaints. Auto manufacturers spent years engineering quieter vehicles, and have created cars whose approach is so subtle that they pose a danger to blind pedestrians. But while engine noise has decreased and car alarms are less reactive, horn honking that is linked with remote keyless entry (RKE) technology increasingly contributes to community noise. RKE horn noise has never been the subject of public health inquiry. In scientific literature, discussion of road noise and health does not distinguish noise among separate sources, and tends to measure aggregate ambient noise levels rather than impulsive noise. RKE horn noise violates state traffic laws and some local noise ordinances regarding horn use, but there have been no legislative attempts to address the technology. This raises questions about whether political leaders and policy setters are not exposed to RKE noise or do not discern RKE sounds from traffic noise, and are therefore unaware of it. Using available auto industry data and case studies, this paper will introduce key facts about RKE horn use in the United States and Canada, reviewing new technologies that render noisy counterparts still in use as redundant.

3pNSc2. Determination of noise emission data of construction sites. Ilya E. Tsukernikov, Igor L. Shubin (Acoust. Lab., Res. Inst. of Bldg. Phys., Odoevskogo proezd, h.7, korp.2, fl. 179, 21 Lokomotovny pr., Moscow 117574, Russian Federation, 3342488@mail.ru), Nikolay I. Ivanov (Dept. of Health and Safety, Baltic State Tech. Univ., Sankt-Petersburg, Russian Federation), Tatiana O. Nevenchannaya (Dept. of Phys., Moscow State Univ. of Printing Arts, Moscow, Russian Federation), and Igor A. Nekrasov (Stock Co. Algoritm-Acoustics, Moscow, Russian Federation)

The reasons for development and substantive provisions of Russian standard GOST R 53695-2009 "Noise. Method for determination of noise emission data of construction sites" are presented. The concept of the noise emission characteristics of a construction site and the method of their determination are entered. Various kinds of civil work, construction site location, environment acoustic conditions, and feature of a landscape are taken into account. Instead of noise of separate sources operating inside a construction site noise of the construction site as a whole sound source is considered. The mean time averaged and maximum values of A-weighted sound pressure levels along the sides of a construction site are taken as its noise emission data. Sound reflection from the barriers located near to a construction site is considered by means of environmental correction for which determination the special method was developed. Procedures of determination of measurement uncertainty, the noise characteristics declared values which are brought in construction site specifications and accuracy degree of the method to be applied are considered.

3pNSc3. Engine sounds of small boats at night transmitted to room in apartment built along canal. Kenji Muto and Toru Akahira (Commun. Eng., Shibaura Inst. of Technol., Toyosu 3-7-5, Koto-ku, Tokyo 1358543, Japan, k-muto@shibaura-it.ac.jp)

In this paper, we show the measurement results of the engine sounds of small boats that cruise in a canal. The canal that is called an Toyosu canal is in the residential area in Tokyo in Japan. It is a canal with the role of the waterway traffic in Tokyo. The engine sounds were measured there from October to November in 2011. There were a lot of tugboats or tugboats pulling a freighter in daytime. There were a lot of fishing boats day and night, and there were a lot of houseboats at night. Many of engine sounds were sound exposure level around Lae = 80 dB. The level of the the greatest was more than sound exposure level Lae = 90 dB. Especially, the most of boats were passed in the morning and evening. They passed in the early morning when the background noise was 50 dB. They passed while obstructing the conversation or waking up. It was a sound with the influence for the inhabitant by the canal. It was shown the result of the engine sound transmitted to the room. The engine noise of the boat was transmitted to the canal side room with high level. These results was described in this paper.

3pNSc4. Aerodynamic noise reduction of a gangway in a high-speed train. Hee-Min Noh and Hyo-In Koh (Korea Railroad Res. Inst., #176, Cheldo bangmulgwan-ro, Uiwang 437-757, South Korea, hmnoh@krri.re.kr)

Excessive interior noise of high-speed trains causes annoyance, fatigue, and stress to passengers. Moreover, the noise occurred in gangway is greater than other noise in the room. Therefore, a research for gangway noise reduction was carried out. At first, cavity noise which causes major noise between car-sections was simulated with FLUENT 6.0 (computer fluid dynamics program). From the simulation result, the flow feedback loop phenomena in the cavity were observed. Then, noise measurements at internal and external positions in between-cars sections were conducted during the driving of a high-speed train. From the measurement results, noise characteristics of gangway and between-cars section were identified. Finally, noise mitigation methods were suggested in this paper.

3pNSc5. Improvement of the acoustic environment inside the high-speed train stations depending on the increase of the speed. Chan Hoon Haan and Chan Jae Park (Architectural Eng., Chungbuk National Univ., 52 NaeSudong-Ro, HeungDeok-gu, Chungbuk National Univ., CheongJu, Chungbuk 361763, South Korea, chhaan@chungbuk.ac.kr)

The speed of trains has been increased due to the development of railway technologies. Recently, operation speed up to 400 km/h is come to effect in Korea. But, it can be easily predicted that noise and vibration could be increased depending on the speed of trains. Especially, train stations are exposed to much noises for 24-h at the nearest place when high-speed trains stop or pass the terminals. In the present study, noise levels of the passing high-speed trains were measured in four different stations and noise levels at the speed up to 400 km/h were calculated. Also, the predicted noises were analyzed and compared with the interior noise criteria (NC-curve). As a result, it was found that the noise levels exceed 10 dB higher than the noise standards in average when train speed was 350 km/h. Based on the results, some design proposals are suggested to satisfy with the noise standards including reinforcement of walls and ceilings, change of finishing materials, which can improve the sound insulation of rooms in the train stations.

3pNSc6. Position optimization of Helmholtz resonator in ducts using a genetic algorithm. Maria A. Nunes and Gabriela Silva (Faculdade UnB Gama - Automotive Eng., Universidade de Brasília, Área Especial de Indústria Projeção A – UnB Setor Leste, Gama, DF 72.444-240, Brazil, maanunes@unb.br)

The Helmholtz resonators (HR) are classic reactive mufflers used to attenuate noise at low frequencies mainly pure tones propagating in ducts from venting systems. In industrial environments the equipments layout, the maintenance and operation purposes limit the installation of this kind of device in terms of space and location. As part of the muffler design it is necessary to considerate these restrictions and an optimization process may necessary in order to increase its acoustic performance. Keep in mind that in real application the downstream radiating end of the duct must be modeled as an open end radiating into free space, the insertion loss (IL) parameter is more proper for evaluating the HR's performance than the transmission loss. Using the IL to estimate the effectiveness of the acoustic filter, the main purpose of this paper is to numerically analyze and maximize this parameter in the maximum attenuation frequency considering position restrictions (bounds constraints) in a duct. An evolutionary search algorithm (GA) has been applied in order to solve the best position for a fixed shape HR in a duct. The finite element method was used to model the acoustic system HR/duct. The pressure data and the optimization step were processed in MATLAB®. For optimal positions the results reveal an increase of 19 dB in the IL parameter at the desired frequency. To verify the sensibility of the methodology simulations were performed varying some GA parameters.

3pNSc7. Possibility of sound environmental design by introducing wave sound into the indoor space. Takane Terashima (Architecture, Mie Univ., 1577 Kurimamachiya-cho, Tsu 5148507, Japan, tera@arch.mie-u.ac.jp) and Kazuhiro Shimahashi (School of Design & Architecture, Nagoya City Univ., Nagoya, Japan)

The purpose of this study is to develop the means of designing sound environment of waterfront area. As one of the means of improving sound environment in the campus space of our university, introducing wave sound from the adjoining seashore into campus area has been proposed and its feasibility have been studied. But wave sound reaches 300 m inland at most and cannot be listened in the most of campus area outdoor. So we plan to introduce wave sound by picking up through microphone and steaming over the campus LAN. In this report, if wave sound is streamed and broadcasted to indoor spaces, the influence of wave sound as background/masking noise on the mental state of users in the space is studied. The samples of various wave sound recorded at seashore near the campus are broadcasted in indoor spaces, such as cafeteria, learning spaces, etc. And subjects are asked to answer the questionnaire about preference and subjective impression for indoor environment. The results show that wave sound is almost accepted by users, but to be recognized by uses, output level must be high and could be harmful. The optimum level of wave sound in the space is discussed.

3pNSc8. A design of control signal in reducing discomfort of the dental treatment sound based on auditory masking. Yuko Suhara, Daisuke Ikefuji, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is023083@ed.ritsumei.ac.jp)

In dental treatment, patients feel a strong discomfort feeling by the treatment sounds which arise by a tooth grinding. In order to add comfort to quality of life, we aim to reduce the discomfort feeling with dental treatment sounds. We previously proposed the unpleasantness reduction method based

on auditory masking to reduce discomfort feeling of noise. The previously proposed method can reduce discomfort feeling by emitting a control signal to a listener, but we had focused on unpleasantness reduction to noise which has a peak frequency. Meanwhile, dental treatment sounds tend to consist of multiple spectral peaks. Therefore, in the present paper, we propose the design method of control signals for reducing discomfort feeling of dental treatment sounds which have multiple spectral peaks. More specifically, we detect the main spectral peaks, which bring a discomfort feeling, and design the control signal, which can mask these spectral peaks. Also, we employ the sound of running water as a source for the control signal. We carried out subjective evaluation experiments to confirm the effectiveness of the proposed method. As a result of evaluation experiments, we confirmed the effectiveness of the proposed method.

3pNSc9. Noise in hospitals as a strain for the medical staff. Silvester Siegmann and Gert Notbohm (Inst. for Occupational and Social Medicine, Heinrich Heine Univ. Duesseldorf, Universitätsstr. 1, Duesseldorf, NRW D-40225, Germany, siegmann@uni-duesseldorf.de)

Noise research in hospitals focuses mainly on the harmful effects on patients. But at least in intensive care units and operation theaters, also the staff is exposed to high levels of noise during considerable portions of working time. Evidence from literature is summarized here. During operation sessions lasting from 30 min to several hours, reported average Leq values range from 58 to 72 dB(A) with maximum levels above 105 dB(A). Similar noise levels are reported from emergency departments. As concentration, precise communication, and fast decisions are necessary in these situations, the acoustical environment has to be considered an enormous strain for the staff and a potential risk with regard to faults at work. But also during normal day and night shifts in intensive care units, noise is mentioned as an important disturbance by the medical staff. Most disturbing are noises from telephones and other communication tools and the signals and sounds from medical devices. Questionnaire surveys result in 80 to 91% of the staff reporting negative effects of noise in their daily work. A variety of measures for noise reduction and prevention in hospitals is suggested in literature emphasizing that the staff plays a decisive role in such projects.

3pNSc10. A consideration on sound masking system for achieving speech privacy using parametric acoustic array speaker. Takahiro Tamesue and Tetsuro Saeki (Yamaguchi Univ., 2-16-1, Tokiwadai, Ube 755-8611, Japan, tamesue@yamaguchi-u.ac.jp)

Speech privacy in open spaces is becoming increasingly important in various situations. Although measures such as the use of sound partitions are already used in many cases, measures that mask speech by emitting sounds have also been considered. A method of masking meaningful speech with meaningless noise would be valuable. Because of this, previous studies have investigated the ability of meaningless steady noise to mask speech and consequently achieve speech privacy. However, the research to date has focused on evaluating speech privacy when the masking noise is emitted from the normal loud speaker system all over the room. The masking noise emitted to the area where high level of speech privacy is not required, may cause an increased psychological impression of annoyance, leading to a decline in performance. In this study, we used a highly directional sound from modulated ultrasound as a masking noise for achieving speech privacy in the narrow area. Psychological experiments were conducted in which the masking sound was transmitted to participants from frontal or above directions with a parametric acoustic array speaker. Using the experimental data, the relationships between the degree of speech privacy and frequency characteristics and directivity of parametric acoustic array speaker were investigated. The results suggested that it is possible to maintain speech privacy in the narrow area by presenting highly directional masking sound.

3pNSc11. Categorization of street types in urban thermoacoustic analysis. Elcione L. Moraes, Irving M. Franco, Marcelle V. Silva, Isabela A. Rocha, Dorival F. Pinheiro, and Mindiyarauakti P. Freitas (Architecture and Planning, Federal Univ. of Pará, av. Augusto Corres, 01, Belém, Pará 66075900, Brazil, elcione@ufpa.br)

Urban areas suffer several environmental perturbations as a result of human activities and technological developments that contribute to the formation of heat islands and increasing noise contamination. Environmental effects are incorporated by population in urban areas and, especially, in areas near roads with heavy traffic. This paper presents a theoretical-experimental analysis on the relationship between climatological conditions and the propagation of noise in traffic corridors with high, medium, and low intensity. Some variables, such as the width of the streets, the height of the buildings, the distance between buildings, the volume flow of vehicles offer the possibility to make traditional techniques for mitigating the air temperature increase and reduce noise levels in urban zones. The results obtained in this work by measuring temperature, humidity, and noise levels, made during certain periods of time in different parts of the city of Belem/Brazil, were linked to a database georeferenced (GIS) that allowed interpolation of data in a single platform, enabling integration between data allowing to correlate them in order to assess which typological conditions are most favorable to the thermoacoustic comfort.

3pNSc12. Effects of age on feasible sound level of possible warning sounds for quiet vehicles. Katsuya Yamauchi (Faculty of Eng., Nagasaki Univ., Bunkyo 1-14, Nagasaki 852-8521, Japan, yamauchi@cis.nagasaki-u.ac.jp), Takayuki Shiizu, Fumio Tamura, and Yuichiro Takeda (Pioneer Corp., Kawagoe, Japan)

It has been noted that reduced noise can also lead to potentially dangerous situations for pedestrians because electric and hybrid vehicles are quieter than conventional internal combustion engine vehicles. Hence, the use of warning sounds which are radiated by the vehicle to alert pedestrians is being considered by various governments. To design the sound itself or to develop the regulation concerning the sound, it is much important to know the feasible sound level of the warning sounds compared to the background sounds. Pilot studies on this topic were performed by Yamauchi *et al*. in 2011 with young subjects. This present study was aimed to reveal the effect of age on feasible sound level of warning sounds. In the experiment, level of five possible warning sounds was adjusted in three different urban environmental sounds in a laboratory. Thirty subjects aged from 19 to 74 years old participated in the experiment. The subjects were asked to adjust the level of warning sounds so that they are clearly audible or just audible depends on the instruction. Results of the adjustments are presented and compared to current recommendations for sound levels of warning signals in quiet vehicles.

3pNSc13. Green cork-based innovative resilient and insulating materials: Acoustic, thermal, and mechanical characterization. Marco Caniato, Sbaizero Orfeo (Architecture and Eng., Univ. of Trieste, via valerio 6/a, Trieste 34100, Italy, mcaniato@units.it), Jan Kaspar, and Roberta Di Monte (Dept. of Chemistry Sci., Univ. of Trieste, Trieste, Italy)

Nowadays, efficient thermal insulation is a principal requirement for buildings and, accordingly, huge amounts of insulators are applied in the constructions, particularly for external walls, radiant floor, etc. Acoustic insulation is another of the most stringent parameters to be taken into account both in the construction of new buildings or their rejuvenation in order to obtain good internal comfort. On the other hand, the use of bio-derived construction materials is gaining stronger and stronger interest. Cork has a low density (120–240 kg m^3) and can be regarded as a hydrophobic and viscoelastic material, with good thermal and acoustic insulation properties. With respect to wood, it presents good resistance to microbial activity and water. Last but not least is the negative carbon fingerprint of cork-based materials. Here we will describe a new class of polymer—inorganic oxides—cork composites that feature enhanced thermal and acoustic properties with respect to traditional commercial composites and maintain, at the same time, all the favorable properties of conventional cork-base composites.

3pNSc14. Impulse response measurement in public space using musical signal including swept-sine signals. Fumiaki Satoh, Junichi Mori, Tomoya Nishii, and Hideki Tachibana (Chiba Inst. of Technol., Tsudanuma 2-17-1, Narashino 275-0016, Japan, fumiaki.satoh@it-chiba.ac.jp)

In design of public spaces, e.g., railway stations, airport terminal buildings, and underground shopping centers, careful attention should be paid from an acoustical viewpoint. It is not only for acoustical comfort but also for safety ensured by a public address system with high intelligibility. As a study for this aim, we have been investigating acoustical characteristics of various public spaces. In these studies, it is strongly desired to measure impulse responses in the spaces under live condition with occupants but the

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usual measurement method using Swept-Sine signals are not applicable because the signals sound very peculiar to the occupants. To mitigate such a problem, we are trying a method using test music signals in which Swept-Sine signals are inserted. In this paper, the availability of this measurement technique is outlined and some measurement results are introduced.

3pNSc15. Modeling room impulse response via composites of spatial-temporal Gaussian processes. Tatsuya Komatsu (Grad. School of Information Sci., Nagoya Univ., Furo-cho, chikusa-ku, Nagoya, Aichi 464-8601, Japan, komatsu.tatsuya@g.sp.m.is.nagoya-u.ac.jp), Gareth W. Peters (Dept. of Statistical Sci., UCL, London, United Kingdom), Tomoko Matsui (Dept. of Statistical Modeling, Inst. of Statistical Mathematics, Tokyo, Japan), Ido Nevat (Wireless & Networking Lab., CSIRO, Sydney, NSW, Australia), and Kazuya Takeda (Grad. School of Information Sci., Nagoya Univ., Nagoya, Japan)

We develop a novel algorithm to estimate a spatial-temporal transfer function of a time-domain room impulse response for reverberation in closed environments. This novel approach involves developing two non-parametric models, one for the early phase and the other for the late phase for reverberation. These models are based on a composite of two Gaussian Process (GP) structures. We also investigate the impact of the choice of spatial and temporal kernels on the estimation and prediction performance. The proposed algorithm incorporates as a special case the widely utilized exponentially decaying model and also extends this model structure within the GP setting to more advanced spatial-temporal structures suitable to perform estimation of the reverberant transfer function. The performance of the proposed algorithm is evaluated in a real environment using nine spatially distributed microphones. The microphones collect reverberant response from a directional speaker allowing observation of noisy realizations of the impulse response to reverberate during early and late stages. We compare the performance of our algorithm with a 3D spatial-temporal cubic interpolation algorithm and show that the proposed algorithm provides equal or better performance than the cubic interpolation.

3pNSc16. Large-scale sound field rendering with graphics processing unit cluster for three-dimensional audio with loudspeaker array. Takao Tsuchiya (Dept. of Information Systems Design, Doshisha Univ., 1-3 Tatara-Miyakodani, Kytanabe City, Kyoto 610-0321, Japan, ttsuchiy@mail.doshisha.ac.jp), Yukio Iwaya (Dept. of Elec. Eng. and Information Technol., Tohoku Gakuin Univ., Tagajo, Miyagi, Japan), and Makoto Otani (Dept. of Comput. Sci. and Eng., Shinshu Univ., Nagano, Nagano, Japan)

The sound field rendering is a technique to simulate the sound field from the three-dimensional numerical models constructed in the computer, and it is the same concept as the graphics rendering in the computer graphics. In this paper, a graphics processing unit (GPU) cluster system is applied to the sound field rendering for a large room simulation. The compact explicit finite difference time domain (CE-FDTD) method is implemented on the GPU cluster system. The CE-FDTD method is a kind of the finite difference method in which the wave equation is directly discretized based on the central differences. The developed GPU cluster system consists of eight PC nodes in which four GPUs are mounted respectively. The rendering results are reproduced by a speaker array system in which 157 speakers surround a small room. The sound field renderings are performed for a large room with a volume capacity of about 5000 cubic meters, in which the impulse responses of one-second length with a sampling rate of 40 kHz are calculated at 157 points corresponding to the loudspeaker positions. The impulse responses are then convoluted with dry music sources. The sound field rendering with the 157ch loudspeaker array system provides the realistic sound field reproduction with natural reverberation.

3pNSc17. Acoustic characterization of three archeological sites in the state of Guanajuato, Mexico. Alejandro Ramos-Amezquita (Comput. Sci. Dept., Tecnológico de Monterrey, Calle del Puente 222, Colonia Ejidos de Huipulco Tlalpan, Mexico City, Mexico DF 14380, Mexico, alejandro.ramos.amezquita@itesm.mx) and David I. Ibarra-Zarate (CAEND, Universidad Politécnica de Madrid, Madrid, Spain)

The present work shows the results obtained in collaboration with the government of the state of Guanajuato in Mexico in a project that looked to include the acoustical analysis of Archaeological sites as a tool for gathering information regarding the historical social use of the areas in question. To that end, the acoustical characterization of 3 archaeological sites recently opened to the public in the state was in order: Cañada de la Virgen, Peralta and Plazuelas. Results include the 3D modeling of the areas of interest and the simulation of the acoustic response of them using the software EASE. Specific acoustic parameters were extracted from the simulations and then analyzed in comparison to archeological hypothesis of the use of such spaces as areas of public appearances, performance, ethno-musicological reports on the type and use of musical instruments, and other archaeological findings in the area in order to support or disprove such hypothesis.

3pNSc18. Basic study on discrimination between sound fields of architectural spaces. Maya Katoh and Takane Terashima (Architecture, Mie Univ., 1577 Kurimamatiya-cho, Tsu, Mie 514-8507, Japan, 412m409@m.mie-u.ac.jp)

Many objective criteria by attenuation property of room acoustic energy have been suggested, and relationships with subjective attributes have been elucidated. The difference of synthetic subjective impression between sound fields could be discriminated by acoustical parameters, i.e., objective criteria above, but details of mechanism in discrimination among different sound fields, weightings or grounds of judgment, etc. have not been clarified. The purpose of this study is to clarify the discrimination factor over difference between sound fields. In this report, subjective experiments are carried out by using impulse response data of existing ten concert halls and music data, made from convolution of impulse responses and dry sources, as stimuli. In these experiments, subjects are asked to evaluate the impression of each stimulus, and to judge the difference among stimuli in pared comparison. The results of these experiments are analyzed and it seems that the factor related to the difference of Reverberance is dominant in discrimination by factor analysis, but there are some cases in which Loudness or Clarity is dominant. The weightings of factors, the boundary to switch judgment between factors, etc. will be discussed.

3pNSc19. Effect of visual information on subjective impression for sound field in architectural space. Yuko Wani, Takane Terashima (Architecture, Mie Univ., 1577 Kurimamachiya-cho, Tsu, Mie 514-8507, Japan, 412m421@m.mie-u.ac.jp), and Yasunobu Tokunaga (Civil Eng., Maizuru National College of Technol., Maizuru, Japan)

In architectural and urban space, we are always exposed to multimodal stimuli of visual information and sound fields in various scenes of everyday life. The purpose of this study is to clarify relationship of subjective impression of vision and auditory, and acquire knowledge which contributes to architectural design or acoustic design. In this report, two following experiments are carried out in which subjects are presented with sound fields by real time convolution as auditory stimuli and panoramic VR images of 360 interactive views of interior as visual stimuli. (1) Comparison between subjective responses for single or multi modal presentations of visual and auditory stimuli from various architectural spaces. (2) Comparison between subjective responses for various combinations of multi modal presentations of visual and auditory stimuli from various architectural spaces. Analysis for results of these experiments clarify the influence of visual information upon the subjective impression for sound field and mutual relationship between subjective impression of vision and auditory of interior of buildings. It is already found that visual information significantly effects subjective impression for sound field by experiment 1, and the details of relationship between elements of visual information and parameters of sound field will be clarified by experiment 2.

3pNSc20. Study of the acoustic of Jean Nouvel's Auditorium 400, at the Museum Reina Sofia in Madrid. Emiliano del Cerro and Silvia M[a] Ortiz (TIC, Universidad Alfonso X el Sabio, Avenida Universidad 1, Madrid 28691, Spain, ecerresc@uax.es)

The Auditorio 400 is one of the buildings that make up the National Art Museum Reina Sofia in Madrid. It is the work of renowned French architect Jean Nouvel. This space was designed to accommodate primarily chamber music concerts but now can be considered as a multi-purpose venue. This hall hosts events with different content: acts with the voice as main sound

source as conferences, seminars etc. and concerts with music from diverse styles, classical, contemporary, avant-garde, and electro acoustic music. This versatility assumes that the acoustic conditions required for the different uses of Auditorio 400 must be diverse and special depending on the sound source, in order to achieve the adequate sound quality for the various events that are held there. This paper presents the study of the acoustics of the Auditorium 400, analyzing various parameters for evaluating the sound quality of the room, highlighting the worst areas of listening, the reasons for the existence of such areas and the description of improvements to be made to ensure that the enclosure meets the expectations in a hall of its relevance.

3pNSc21. Influence of visual information on sound evaluation in auditorium. Yasunobu Tokunaga, Daichi Okuie (Maizuru National College of Technol., 234 Shiroya, Maizuru-shi 625-8511, Japan, tokunaga@maizuru-ct.ac.jp), and Takane Terashima (Grad. School of Eng., Mie Univ., Tsu-shi, Japan)

When hearing music in an auditorium, audience is provided with aural information and visual information at the same time. Visual and auditory sense sometimes interact with each other so that the auditory sense is considered to have some influence on sound evaluation made by audience. The purpose of this study is to reveal an influence of visual information obtained in audience seats in a hall on subjective evaluation. We investigated the relationship between sound evaluation and whether a musical performance video as visual stimulus is provided or not, and between the sound evaluation and evaluation concerning the space at certain position in the audience seats. As a result, it was revealed that visual information gave statistically significant influence on sound evaluation.

3pNSc22. Design and positional accuracy of straight-path traversing room acoustic measurement system based upon low-cost servo motor and light-weight multi-field microphone. Roger M. Ellingson (RM Ellingson Design & Development, LLC, 8515 SW Barnes Rd., Portland, OR 97225, Rogere@Rmegen.net) and Guillaume J. Bock (Bruel & Kjaer, North America Inc., Snohomish, WA)

The construction and operation of a straight-path traversing, acoustical microphone-based, measurement system is described. The system was designed to support test room qualification procedures such as prescribed in standards ANSI S12.35-1990 and Annex A of ISO 3745-2003. Major system components include a taut line suspending a sliding microphone carriage drawn by a string attached to a rotating drum. Central to the overall light-weight, low-power, mechanical design is the physically small Bruel & Kjaer 4961 microphone, holder, preamp, and signal cable whose multi-field response characteristic should well support accurate room qualification measurements. The battery-powered drum drive mechanism is built using the servo motor, wheels, and gears from a Lego Mindstorms NXT kit. Precision motor rotation is remotely programmable over the NXT controller's wireless Bluetooth interface. Commonly available sport fishing tackle composes the majority of the suspension assembly. A software interface library is also described which enables PC-based applications to automate microphone positioning in synchrony with source emission, signal acquisition, and analysis. The system has been used to document the free-field characteristic of a perimeter loudspeaker array with centrally located listener in a fully anechoic chamber environment. Together with construction and operational detail, results indicating overall microphone positioning accuracy and reliability are presented here.

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 519B, 1:00 P.M. TO 2:40 P.M.

Session 3pPAa

Physical Acoustics: Borehole Acoustics Logging for Hydrocarbon Reservoir Characterization II

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Contributed Papers

1:00

3pPAa1. Simulation of sonic logging for deviated wells in anisotropic formations. Evgeniya Deger (SKK, Schlumberger, 2-2-1 Fuchinobe, Chuo-Ku, Sagamihara, Kanagawa 252-0206, Japan, emyalo@slb.com), Marwan Charara (SMR, Schlumberger, Moscow, Russian Federation), Henri-Pierre Valero (SKK, Schlumberger, Sagamihara, Japan), Denis Sabitov, and Grigory Pekar (SMR, Schlumberger, Moscow, Russian Federation)

Interpretation of sonic logging data acquired in environments with complex anisotropy is a difficult problem attracting attention of researchers and oil industry. In order to better understand physics of wave propagation in highly anisotropic medium and be able explaining observations from field data there is a need for fast and accurate numerical modeling capability. To address such challenge, we developed an efficient and accurate numerical algorithm for the simulation of sonic logging experiments in highly anisotropic formation. The basis of the approach is a heterogeneous spectral element method implemented on multi-GPU applied to acoustic-elastic wave equation. The approach was designed to simulate wave propagation in 3D arbitrary anisotropic elastic media with attenuation for a constant quality factor via standard linear solid using the tau-method. Due to the use of an unstructured grid, the spectral element algorithm enables handling tools in a fluid-filled borehole with surrounding geological models of high complexity. Several examples of log simulations for deviated wells in VTI formations for monopole, dipole, and quadrupole source symmetries and their comparison with real field data will be presented in this paper. Discussion regarding complex wave propagation will be developed in view of these simulations and real data.

1:20

3pPAa2. Numerical simulations of dipole sonic responses in a liquid-filled trough with arc-shaped section. Xiao He, Hao Chen, and Xiuming Wang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., 21, Northern 4th Ring Rd. West, Haidian District, Beijing 100190, China, hex@mail.ioa.ac.cn)

To test the running performance of a sonic logging tool, it is an effective way to place the tool in a water-filled trough and record the sonic responses for an integrated check of the transducers in laboratory. Hence it is necessary to investigate the wave propagation in such a non-symmetric structure. In this study, we present the numerical modeling results of dipole sonic responses in a trough with arc-shaped section. A 3D cylindrical finite

difference code as well as the irregular free-surface conditions is implemented. It is revealed that the flexural mode in a trough is evidently slower than that in a cylindrical pipe with same sizes. The flexural velocity decreases with the increasing gap angle of the trough. Moreover, the trough structure shows strong elastic anisotropy. The in-line responses (XX and YY) of the logging tool are difference in phases and amplitudes. The amplitudes of both cross-line responses (XY and YX), which reflect the level of anisotropy, become greater as the gap angle increases. The waveforms with varying dipole orientations are also illustrated. The XY and YX responses excited by an inclined transmitter are much stronger than those generated by a horizontal or a vertical dipole source.

1:40

3pPAa3. Research on a kind of low-frequency broadband cross-dipole projector. Dai Yuyu, Xin Penglai, Wang Xiuming, and He Hongbin (The Ultrasonic Phys. and Exploration Lab., Inst. of Acoust., Chinese Acad. of Sci., No.21, Bei-Si-huan-Xi Rd., Beijing 100190, China, daiyuyu001@126.com)

The finite element method (FEM) is used to simulate a low-frequency broadband cross-dipole projector based on trilaminar bender bar in this paper. In the simulated model, four long trilaminar bender bars and four short trilaminar bender bars are attached on a skeleton two form two square arrays, and very array excite two different response frequencies. The four response frequencies distribute on the range from 400 Hz to 5 kHz to reach broadband exciting. Finally, a sample is fabricated, and test in an anechoic tank, it is shown that the test result meet the simulated result very well.

2:00

3pPAa4. Phase-based dispersion analysis of borehole acoustic logs. Said Assous (Geoscience, Weatherford, East Leake, Loughborough LE126JX, United Kingdom, said.assous@eu.weatherford.com), Laurie Linnett (Geoscience, Weatherford, Edinbourgh, United Kingdom), and Peter Elkington (Geoscience, Weatherford, East Leake, United Kingdom)

The dispersive behavior of acoustic waves in boreholes is of interest in the evaluation of reservoir rocks, particularly from the point of view of near wellbore stress distribution. It is also used as a quality control on dipole sonic calculations that estimate formation shear slowness from the low frequency asymptote of the flexural wave slowness. Multiple methods are available for dispersion analysis; the paper reviews the most commonly used, including the Prony and the spectral semblance methods, and proposes a new phase-based analysis technique that has the benefit of improved slowness resolution. The methods are applied to synthetic and real data sets, and results compared. The new method is show to have lower slowness uncertainty for any given frequency, and the upper and lower frequency limits for which dispersion can be calculated is also extended.

2:20

3pPAa5. Unrelaxed drained bulk modulus for fluid-saturated rocks on full frequency range. Yong J. Song, Hengshan Hu (Dept. of Astronautics and Mech., Harbin Inst. of Technol., P.O. Box 344 92 West Dazhi St., Harbin, Heilongjiang (+0086)150001, China, songyongjia061220110@126.com), and Changwen Li (Technol. Ctr. of China Petroleum Logging CO., LTD., Xian, China)

Local flowing between cracks and pores is called squirt-flow that usually induce elastic moduli dispersion and waves attenuation. Expression of unrelaxed drained bulk modulus on full frequency range is derivated in this paper when liquid pressure in soft crack is unrelaxed. Unrelaxed drained bulk modulus's real part increases with frequency, and unrelaxed drained bulk modulus's imaginary part is nonzero. This studies also show that liquid pressure in cracks equals to zero at he low frequency limitation, that is to say liquid pressure in cracks have sufficient time to relax and in this case the drained bulk modulus correspondents to Biot's static drained bulk modulus.At the high frequency limitation, the unrelaxed drained bulk modulus approximate to Mavko-Jizba's expression. The expression of drained modulus in this paper also degenerates to Biot theory's drained bulk modulus when crack density equals zero, but the latter is just a modulus on a hypothetical state and is not the true static experiment data when rocks contain soft cracks. Unrelaxed drained bulk modulus is used to calculate fast P-wave and slow P-wave velocities and attenuation instead of static drained or dry bulk modulus in Biot's theory. Squirt-flow generates much more velocity dispersion and attenuation in fast P-wave than Biot flow. The magnitude of attenuation depends on crack density and the relaxation frequency depends on aspect ratio.

WEDNESDAY AFTERNOON, 5 JUNE 2013 519A, 1:00 P.M. TO 2:40 P.M.

Session 3pPAb

Physical Acoustics: General Physical Acoustics II

Raymond Panneton, Chair
Mech. Eng., Univ. de Sherbrooke, 2500 Universite Blvd., QC J1K 2R1, Canada

Contributed Papers

1:00

3pPAb1. Modeling sound fields from radially symmetric impulsive planar sources using Rayleigh's Integral. Stephen I. Warshaw (LLNL (Ret.), Univ. of CA, 40 West 15 St Loft 1C, New York, New York 10011, siw1939@yahoo.com)

The evaluation of Rayleigh's integral in cases where the integrand contains delta functions with time-dependent arguments provides a highly useful and easily implemented vehicle for modeling and understanding the radiation of impulsive sound waves from planar sources. We present elementary examples of such calculations and show their application in various realistic scenarios, including the radiation of impulsive aeroacoustic sound from buried explosions and potential extensions to lithotripsy. In this paper, we represent the surface motions of a planar radiating source as delta functions of radially expanding or contracting arguments with easily specified geometrical parameters, and present model near-field to far-field calculations of the launched sound waves using analytic and numerical integration. We show that significant features in these traveling impulses can easily be related to the details of the kinematic history of the planar source. We restrict our purview to the time domain and to radial symmetry; frequency analyses and azimuthally asymmetric motions will not be considered.

1:20

3pPAb2. Application of the spectral method for computation of the spectrum of anisotropic waveguides. Timur Zharnikov, Denis Syresin (Schlumberger Moscow Res., Pudovkina St., 13, Moscow 119285, Russian Federation, tzharnikov@slb.com), and Chaur-Jian Hsu (Schlumberger-Doll Res., Cambridge, MA)

Spectral method is formulated in cylindrical coordinates for the general case of waveguide with arbitrary anisotropy with the spatial dependence. According to the idea of this approach, matrix representation of operator in

the right-hand side of governing equations is considered. As a result, the latter are cast into exact infinite set of integro-differential equations. Explicit expressions for their kernels expose coupling between axial and azimuthal harmonics. Coupling of axial harmonics vanishes in important case of waveguide with translational invariance in axial direction. It results in the set of differential equations, which is used to introduce practical approximation procedure. The latter yields generalized eigenvalue problem, which can be solved numerically for the spectrum of the operator. The spectrum is sorted according to eigenmodes' properties. Thus dispersion curves of eigenmodes are constructed. Presented consideration can be adapted for waveguides of different physical nature (elastic, electromagnetic, etc.) and different geometry (rectangular, elliptical, etc.). Developed technique is verified by comparison with results of controlled laboratory measurements on anisotropic sample. Monopole, dipole and quadrupole normal modes for scaled borehole in anisotropic rock sample with TTI symmetry are considered. The comparison of spectral method results with the dispersion analysis of synthetic data is provided as well.

1:40

3pPAb3. Influence of reflecting walls on edge diffraction simulation in geometrical acoustics. Alexander Pohl (HafenCity Universität Hamburg, Hebebrandstrasse 1, Hamburg 22297, Germany, alexander.pohl@hcu-hamburg.de), Dirk Schröder (EPFL Lausanne, Lausanne, Switzerland), Uwe M. Stephenson (HafenCity Universität Hamburg, Hamburg, Germany), and U. Peter Svensson (NTNU Trondheim, Trondheim, Norway)

Edge diffraction can be introduced into geometrical acoustics mainly by three models: detour-based, energetic and wave-based diffraction models. In the past, we thoroughly compared Maekawa's detour law, the uncertainty relation based diffraction method and the secondary source model by the example of edge diffraction of a single wedge. However, the influence of the wedge shape has not yet been analyzed. Therefore, we consequently study in this contribution the influence of the wedge's faces. This is analyzed by varying both the faces' reflection properties and their opening angle. This is extended to the crucial case of approximately parallel faces (inner angle, e.g., 179°), where diffraction is physically neglectable, but computationally problematic for the uncertainty based diffraction method. Additionally, wedges are placed on an infinitely long surface. Therewith, we can analyze the floor reflections' impact on the sound field behind the wedge by varying both their absorption and scattering coefficients. Furthermore, we discuss artifacts which can arise in the uncertainty based diffraction model due to arbitrary positioning of diffraction planes, so called "transparent walls." Finally, we discuss the advantages and disadvantages of the presented methods.

2:00

3pPAb4. Acoustic response of a buried landmine with a low grazing-angle source array, focused on the ground. Benjamin J. Copenhaver, Justin D. Gorhum, Charles M. Slack, Martin Barlett, Thomas G. Muir, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, bcopenhaver@utexas.edu)

An array of 16 loudspeakers, deployed along a segment of the base of a right circular cone, was used to focus intense tone bursts at low audio frequencies on a soil, with and without a buried target, having a compliant lid. The response of the target site was examined as a function of source frequency, intensity level, and excitation signal type, including multi-tone radiations. Nonlinear interaction to produce sum and difference frequencies at the target site was examined and compared with observations of Korman and Sabatier [J. Acoust. Soc. Am. **116**, 3354 (2004)]. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

2:20

3pPAb5. The farfield impulse response for a rectangular piston in viscous media. Pedro C. Nariyoshi and Robert J. McGough (Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48864, mcgough@egr.msu.edu)

Calculations of the transient radiation pattern in the farfield of rectangular transducers often employ the impulse response of the velocity potential. Analytical expressions for the impulse response are known for lossless media, and similar expressions are needed for propagation in viscous media. Solutions are obtained with two different Green's functions for viscous media. One solution, which is causal, is based on an approximate Green's function for the Stokes wave equation, and the other solution, which is noncausal, is based on an approximate Green's function for the Blackstock wave equation. The impulse response is calculated with these Green's functions using the Rayleigh-Sommerfeld integral, and the results of these calculations are compared to analytical expressions for the impulse response derived for viscous media. Numerical results are obtained for a 1 mm by 1 mm square transducer evaluated in the farfield region, where the Rayleigh-Sommerfeld integral provides the reference calculated with 400 abscissas in each direction. The results show that the causal and noncausal solutions are nearly identical in the farfield region, and the analytical impulse response expressions derived from the causal and noncausal Green's functions are consistently within 1% of the Rayleigh-Sommerfeld reference in the farfield. [Work supported in part by NIH Grant R01 EB012079.]

Session 3pPP

Psychological and Physiological Acoustics: Multimodal Influences on Auditory Spatial Perception

William L. Martens, Cochair
Univ. of Sydney, 148 City Rd., Wilkinson Bldg. G04, NSW 2006, Australia

Shuichi Sakamoto, Cochair
Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai 980-8577, Japan

Invited Papers

1:00

3pPP1. Spatial sound and its effect on visual quality perception and task performance within a virtual environment. Brent Cowan (Business and Information Technol., Univ. of Ontario Inst. of Technol., Oshawa, ON, Canada), David Rojas (SickKids Learning Inst., The Hospital for Sick Children, Oshawa, ON, Canada), Bill Kapralos (Business and Information Technol., Univ. of Ontario Inst. of Technol., 2000 Simcoe St. North, Oshawa, ON L1H 7K4, Canada, bill.kapralos@uoit.ca), Karen Collins (The Games Inst., Univ. ofWaterloo, Waterloo, ON, Canada), and Adam Dubrowski (SickKids Learning Inst., The Hospital for Sick Children, Toronto, ON, Canada)

Immersive 3D virtual environments such as simulations and serious games for education and training are typically multimodal, incorporating at the very least both visual and auditory cues, each of which may require considerable computational resources, particularly if high fidelity environments are sought. It is widely accepted that sound can influence the other modalities. Our own previous work has shown that sound cues (both contextual and non-contextual with respect to the visual scene) can either increase or decrease (depending on the sound) visual fidelity (quality) perception in addition to the time required to complete a simple task (task completion time) within a virtual environment. However, despite the importance and benefits of spatial sound (sound that goes far beyond traditional stereo and surround sound techniques, allowing users to perceive the position of a sound source at an arbitrary position in three-dimensional space), our previous work did not consider spatial sound cues. Here we will build upon our previous work by describing the results of an experiment that will be conducted to examine visual fidelity (quality) perception and task performance in the presence of various spatial sound cues including acoustical reverberation and occlusion/diffraction effects, while completing a simple task within a virtual environment.

1:20

3pPP2. Touch the sound: The role of audio-tactile and audio-proprioceptive interaction on the spatial orientation in virtual scenes. M. Ercan Altinsoy and Maik Stamm (Chair of Commun. Acoust., Dresden Univ. of Technol., Helmholtzstr. 18, Dresden 01062, Germany, ercan.altinsoy@tu-dresden.de)

Being able to localize objects in the space close to the body is an important prerequisite for precise object interaction. It is also very important for the spatial orientation in virtual scenes. Since sound is usually produced by the vibrations of a body, sound emitting objects, such as shaver or hair dryer, provide both auditory and haptic information. This study focuses on auditory-haptic localization in the spatial domain. We carried out two experiments to investigate the interaction effects. In the first experiment, the influence of tactile signals on auditory localization task was investigated. Similar to the ventriloquist effect from auditory-visual interaction, the results of the first experiment show that the perceived location of auditory stimuli is influenced by tactile stimulation. The results also indicate some hints that there may be an audiotactile precedence effect. In the second experiment, the influence of auditory signals on proprioception was investigated. The results show that the auditory and proprioceptive information can be combined in such a way that the localization errors in a virtual scene are minimized.

1:40

3pPP3. Compression of perceived auditory space during forward self-motion. Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Wataru Teramoto (Grad. School of Arts and Letters, Tohoku Univ., Sendai, Japan), Fumimasa Furune (Grad. School of Information Sci., Tohoku Univ., Sendai, Japan), Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan), and Jiro Gyoba (Grad. School of Arts and Letters, Tohoku Univ., Sendai, Japan)

Humans can perceive a stable auditory environment and appropriately react to a sound source, even when they are moving. This suggests that the inputs are reinterpreted in the brain, while being integrated with information on the movements. Although several studies have shown the influence of the vestibular semicircular canal signals on auditory localization, it is not clear how auditory space representation is modulated by linear accelerations which are obtained from the macular receptors of the otolith system (utricle and saccule). We investigated the effect of the linear acceleration on auditory space representation. During the forward/backward self-motion, a short noise burst was presented from one of the loudspeakers which were aligned parallel to the motion direction when the listener's coronal plane reached the location of one of the speakers (null point). The results showed that the sound position aligned with the subjective coronal plane was displaced ahead of the null point only during forward self-motion. Moreover, all the sounds that were actually located in the traveling direction were perceived as being biased towards the null point. These results suggest a distortion of perceived auditory space in the direction of movement during forward self-motion.

2:00

3pPP4. Dominance of head-motion-coupled directional cues over other cues during walking depends upon source spectrum. William L. Martens (Faculty of Architecture, Design and Planning, Univ. of Sydney, 148 City Rd., Wilkinson Bldg. G04, Univ. of Sydney, NSW 2006, Australia, william.martens@sydney.edu.au), Shuichi Sakamoto (Res. Inst. of Elec. Commun. and Grad. School of Information Sci., Tohoku Univ., Sendai, Japan), Luis Miranda, and Densil Cabrera (Faculty of Architecture, Design and Planning, Univ. of Sydney, Sydney, NSW, Australia)

Listeners who walk past a continuously presented speech sound source emanating from a fixed spatial position will typically experience veridical perception of source location. If, however, walking listeners are fitted with binaural hearing instruments that allows for the signals reaching their ears to be interchanged, left for right and right for left, the sound source is typically reported to be located in a spatial region that is reversed with respect to all three spatial axes: left for right, front for back, and above for below. This result has been taken as evidence for the relative dominance of dynamic interaural directional cues over the spectral directional cues associated with each listener's own pinnae which should support veridical perception. In order to investigate the relative importance of the spectral energy distribution of the source on the illusory reversals of source location, bursts of broadband noise were presented rather than continuous speech. Under these circumstances, with greater energy in higher frequency bands, the reversals did not readily occur. Therefore, it has been concluded that head-motion-coupled directional cues are likely to dominate spectral cues associated with the filtering effects of the listener's pinnae only for sources containing greater energy at lower frequencies.

2:20

3pPP5. Impact of dynamic binaural signal associated with listener's voluntary movement in auditory spatial perception. Tatsuya Hirahara, Daisuke Yoshisaki, and Daisuke Morikawa (Faculty of Eng., Toyama Prefectural Univ., 5180 Kurokawa, Imizu 939-0398, Japan, hirahara@pu-toyama.ac.jp)

The effect of listener's voluntary movement on the horizontal sound localization was investigated using a binaural recording/reproduction system with TeleHead, a steerable dummy head. Stimuli were static binaural signals recorded with a still dummy-head in head-still condition, dynamic binaural signals recorded with a dummy-head that followed precise or modified listener's head rotation, dynamic binaural signals produced by steering-wheel rotation with listener's hands in head-still condition, and dynamic binaural signals produced by an experimenter in head-still condition. For the static binaural signals, some were localized within the head and the front-back errors often occurred. For the dynamic binaural signals, none of them was localized within the head, and the front-back confusions seldom occurred. Sound images of the dynamic binaural stimuli produced by head rotation were localized out-of head, while those produced by the steering-wheel rotation or by an experimenter were moving around the listener's head. Listeners could judge the orientation of each stimulus more correctly with dynamic binaural signals produced by listener's head or steering-wheel rotation than with static binaural signals and with dynamic binaural signals produced by an experimenter. Results suggest that the dynamic binaural signal associated with listener's voluntary movement play crucial role in sound localization.

2:40

3pPP6. Cue weighting and vestibular mediation of temporal dynamics in sound localization via head rotation. Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., Elborn College, 2262, 1201 Western Rd., London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

Our studies have quantified the salience and weighting of dynamic acoustic cues in sound localization via head rotation. Results support three key findings: (1) low-frequency interaural time-difference (ITD) is the dominant dynamic binaural difference cue; (2) when available, high-frequency spectral cues dominate front/rear localization; and (3) the temporal dynamics of dynamic cue processing are particular to auditory-vestibular integration. ITD dominance is shown indirectly in findings that head movements are highly effective for localizing low-frequency targets but not narrow-band high-frequency targets and that only normal low-frequency hearing is required to localize via dynamic cues. Direct evidence comes from manipulation of dynamic binaural cues in spherical-head simulations lacking spectral cues. If the stimulus provides access to dominant high-frequency spectral cues, location illusions involving head-coupled source motion fail. For low-frequency targets, localization performance improves with increasing head-turn angle, but decreases with increasing velocity such that performance depends primarily on stimulus duration; ~100 ms being required for accurate front/back localization. That duration threshold only applies in dynamic localization tasks, and not in auditory-only tasks involving the same stimuli. Correct spatial interpretation of dynamic acoustic cues appears to require vestibular information about head motion, thus the 100-ms temporal threshold is likely a property of vestibular-auditory integration.

Session 3pSAa

Structural Acoustics and Vibration, Noise, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials II

Yun Jing, Cochair
Mech. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695

Dean Capone, Cochair
Penn State, P.O. Box 30, State College, PA PA

Contributed Papers

1:00

3pSAa1. Broadband transparent periodic acoustic structures. Gregory J. Orris (Acoust. Div., U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, gregory.orris@nrl.navy.mil), Christopher N. Layman, Christina J. Naify (National Res. Council, Washington, DC), Theodore P. Martin, and David C. Calvo (Acoust. Div., U.S. Naval Res. Lab., Washington, DC)

The creation of acoustically transparent materials is of interest for enhanced energy focusing in metamaterial lenses, vibration isolation and structure concealment in underwater environments. It has previously been shown that metal pentamode metamaterials may provide water like behavior yet retain enough shear to provide structural stability. This is achieved through a periodic lattice with sub-wavelength cells. The current talk presents a study on the simulated and experimental behavior of other structural materials that operate similar to pentamode composites, which exhibit transparency in underwater conditions. Specifically, we examine a two-dimensional design composed of a honeycomb arrangement of fluid inclusions in a solid polymer background. Materials, designs and simulations are supported through both band structure calculations and transmission modeling of finite slabs of the device. Experiments are performed in a water-filled test tank and broad frequency behavior is examined. [Work supported by ONR.]

1:20

3pSAa2. Evaluation of an acoustic metamaterial leaky-wave antenna. Christina J. Naify, Christopher N. Layman (National Res. Council, 4555 Overlook Ave. SW, Washington, DC 20375, christina.naify.ctr@nrl.navy.mil), Theodore P. Martin, David Calvo, and Gregory J. Orris (Acoustics, Naval Res. Lab., Washington, DC)

An acoustic projector array, which can be steered between $\pm$ 90 degrees backfire to endfire directions based solely on input frequency, is presented using a combination of transmission line (TL) analysis and negative index metamaterial ideas. An acoustic version of a leaky wave antenna, this TL structure is composed of acoustically loaded membranes (acoustic masses) and open channels (acoustic shunts). This type of TL structure had been shown previously to have broadband negative index behavior below a cutoff frequency, and positive index behavior above the cutoff frequency. By carefully designing the geometry of the acoustic elements, continuous scanning with no acoustic bandgap was achieved. The fast-wave radiation band of the antenna was determined using a lumped acoustic parameter method. Angle of radiation of the acoustic waves out of the acoustic shunts was continually scanned backfire-to-endfire, including broadside. Applications of this antenna structure include both source and sensing technologies. Finite element analyses and acoustic circuit analysis were used to predict the angle of radiation of the antenna which agreed with experimentally obtained results. [Work supported by the Office of Naval Research.]

1:40

3pSAa3. Underwater sound transmission through thin soft elastomers containing arrays of pancake voids: Measurements and modeling. David C. Calvo, Abel L. Thangawng (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, david.calvo@nrl.navy.mil), and Christopher N. Layman (NRC Postdoctoral Fellow, Naval Res. Lab., Washington, DC)

Measurement of underwater sound transmission through thin (~750 microns) layers of the soft elastomer polydimethylsiloxane (PDMS) containing microfabricated arrays of pancake-shaped cavities is presented. Cavities are 120 microns in diameter and 2.5 microns in height with a nominal lattice spacing of 300 microns. A sound transmission minimum is found at 282 kHz which agrees with predictions of a finite-element model of the array and the value for monopole resonance frequency of an air-filled single pancake cavity in unbounded PDMS. This resonance is a factor of 0.62 lower than the null that would occur for spherical cavities of equivalent volume. The width of the minimum is also significantly broader than that which would be obtained with spherical voids. Modeling results incorporate careful measurements of attenuation for both shear and compression waves in PDMS done in a separate effort. Acoustic transmission variation as a function of lattice spacing and the number of layers is discussed. We also present measurements of transmission through PDMS layers featuring randomly positioned (but not overlapping) pancake cavities to evaluate how the lattice constant (or lack thereof) affects sound transmission near the pancake resonance frequency or in higher acoustic bandwidths. [Work sponsored by the Office of Naval Research.]

2:00

3pSAa4. Broadband acoustic metamaterials with electro-magnetically controlled properties. Dimitri Donskoy (Civil, Environ., and Ocean Eng. Dept., Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, ddonskoy@stevens.edu) and Vladimir Malinovsky (Dept. of Phys. and Eng. Phys., Stevens Inst. of Technol., Hoboken, NJ)

The proposed class of acoustic metamaterials utilizes clouds of electrically charged nano or micro particles exposed to an external magnetic field. The particles are also elastically supported or embedded into elastically compliant medium in a way that the designed structure exhibits two resonances: mechanical spring-mass resonance and electro-magnetic cyclotron resonance. It is shown that if the cyclotron frequency is greater than the mechanical resonance frequency, the designed structure could be highly attenuative (40–60 dB) for vibration and sound waves in a very broad frequency range covering low and very low frequencies. The approach opens up wide range of opportunities for design of adaptive acoustic metamaterials by controlling magnetic field and/or electrical charges.

2:20

3pSAa5. Broadband directional ultrasound propagation using sonic crystal and nonlinear medium. Dipen N. Sinha and Cristian Pantea (Mater. Phys. & Applications, Los Alamos National Lab., MPA-11 D429, P.O. Box 1663, Los Alamos, NM 87545, sinha@lanl.gov)

The development of a passive, sonic crystal-based device with unusual properties will be reported. This device combines a 1D sonic crystal, a nonlinear medium, and an acoustic low pass filter to allow broadband ultrasound propagation as a collimated beam for specialized underwater communication. The signal to be transmitted is first amplitude modulated with a high-frequency ultrasonic carrier wave and applied to one side of the device. The device then demodulates this signal and the original low frequency signal appears as a collimated beam on the other side. The sonic crystal provides a band pass acoustic filter through which the high-frequency ultrasonic signal can pass through and the nonlinear medium then demodulates the signal and also generates the low frequency sound beam through the parametric array concept. The low pass filter removes any remaining high frequency components. The device also functions in a unidirectional manner. Design details of the device and experimental data will be presented.

2:40

3pSAa6. Equations for energy characteristics of oscillatory systems with internal (hidden) degrees of freedom and application to acoustic metamaterials. Yuri Bobrovnitskii (Theor. and Appl. Acoust., Blagonravov Mech. Eng. Res. Inst., 4, Griboedov Str., Moscow 101990, Russian Federation, yuri@imash.ac.ru)

General equations are derived for calculating the kinetic and potential energies and other energy characteristics of linear oscillatory NDOF-systems a portion of DOFs of which are internal or inaccessible for measurement and excluded from consideration. The energy characteristics are expressed through parameters pertaining only to the input or accessible DOFs. The equations are based on the certain novel properties of the so-called Shur matrix complement. The theory is applied to calculating the energy characteristics of acoustic metamaterials for which this is still an unsolved problem, especially for those with negative effective density and stiffness. A metamaterial is thought as a medium or periodic structure in which the role of effective inertia and elastic elements is played by sufficiently complex oscillatory systems with internal DOFs. Applying the derived equations to a cell of periodicity of a metamaterial one can obtain the exact values of the needed energy characteristics. The theory is verified in computer simulation and laboratory experiment.

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 512BF, 1:00 P.M. TO 3:00 P.M.

Session 3pSAb

Structural Acoustics and Vibration: Applications in Structural Acoustics and Vibration I

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

Eric E. Ungar, Cochair
Acentech, Inc., 33 Moulton St., Cambridge, MA 02138-1118

Contributed Papers

1:00

3pSAb1. Squeal noise generated by railway disc brakes: Experiments and stability computations on large industrial models. Olivier Chiello (LTE, IFSTTAR, 25 avenue François Mitterrand, Case 24, Bron Cedex F-69675, France, olivier.chiello@ifsttar.fr), Jean-Jacques Sinou (Laboratoire de Tribologie et de Dynamique des Systèmes, Ecole Centrale de Lyon, Ecully, France), Nicolas Vincent (Vibratec, Ecully, France), Guillaume Vermot des Roches (SDTools, Paris, France), Franck Cocheteux, Selim Bellaj (Agence d'Essai Ferroviaire, SNCF, Vitry-sur-Seine, France), and Xavier Lorang (Innovative and Res. Dept., Phys. of Railway System and Passenger Comfort, SNCF, Paris, France)

The squeal noise generated by railway disk brakes is an everyday source of discomfort for the passengers both inside and outside the trains in stations. The development of silent brake components is needed and requires a better characterization and understanding of the phenomenon. This is the aim of the experimental and numerical investigations performed in the framework of the French AcouFren project and presented in this paper. The first part is concerned with the analysis of experimental data coming from bench tests in a lot of braking configurations including different brake pads. In the second part, the measurements are compared with the results of a large FE model of the brake taking into account the mechanical complexity of each component, especially the brake pads. Components models have been previously updated using experimental modal analysis but the whole model is a direct assembling of it, without updating. The assumption of unilateral contact and Coulomb friction at the pad/disc interface is sufficient to destabilize the sliding equilibrium of the brake and lead to self-sustained vibrations. Complex vibrating modes are computed in order to describe and understand the dynamic instabilities.

1:20

3pSAb2. The modeling of wheel squeal in the time domain and its validation. Xiaogang Liu and Paul A. Meehan (School of Mech. and Mining Eng., Univ. of Queensland, St Lucia, Brisbane, QLD QLD 4072, Australia, xiaogang.liu@uq.edu.au)

Wheel squeal is a tonal noise generated when a train negotiates a curve, whose sound pressure level is normally 30 dB above rolling noise. The sound pressure level of wheel squeal has been shown to increase with the angle of attack and rolling speed in both field and laboratory tests, but the causes behind the manner of increase are still unknown. To investigate this, a model in the time domain was developed by integrating the contact mechanics with the vibration of the wheel to demonstrate how the nonlinear friction creep behavior interacts with the wheel vibration. This model simulated the vibration velocity of the testrig wheel at different rolling speed and angle of attack. The results correlate well with the recorded sound pressure level of wheel squeal. The lateral creepage and lateral force in various situations were also simulated. It was found that due to the interaction of wheel vibration with lateral force and lateral creepage the vibration velocity amplitude of the wheel at a high angle of attack and rolling speed is larger. This explains why the sound pressure level of wheel squeal also increases in the same manner. The phenomenon is explained theoretically using the mechanics based model.

1:40

3pSAb3. The role of pad-modes and nonlinearity in instantaneous mode squeal. Sebastian M. Oberst and Joseph C.S. Lai (Acoust. & Vib. Unit, School of Eng. and Information Technol., UNSW Canberra, UNSW Canberra, Northcott Dr. bld 15/117, Canberra, ACT 2600, Australia, s.oberst@adfa.edu.au)

Disc brake squeal is a major source of customer dissatisfaction and related warranty costs for automobile manufacturers. Although mode coupling is recognized as a mechanism often found in squealing brakes, recent research results show that friction induced pad-mode instabilities could be the cause of instantaneous mode squeal reported in the literature. In this paper, the nonlinear characteristics of instantaneous mode squeal initiated by pad-mode instabilities are studied by analyzing phase space plots of vibrations and sound pressure for a numerical model of a pad-on-plate system as the friction coefficient increases. Results show that as the friction coefficient increases from 0.05 to 0.65, attractors of vibration in the phase space transits from limit cycle to quasi-periodic, showing signs of approaching chaotic behavior. It is shown here that the correlation of the sound pressure behavior in the phase-space with structural vibration is crucial to understanding the role of pad modes and nonlinearity in instantaneous mode squeal.

2:00

3pSAb4. Squeak and rattle noise prediction for trimmed door of a car using hybrid statistical energy—Finite element method analysis. Sajjad Beigmoradi (Automotive Eng. Dept., Iran Univ. of Sci. & Technol., No13, Emmami alley, Golzarand Alley, Safdari St. Navab Safavi St, Tehran, Iran, s.beigmorady@gmail.com), Kambiz Jahani (Mech. Eng. Dept., Sharif Univ. of Technol., Tehran, Iran), and Hassan Hajabdollahi (Mech. Eng. Dept., Iran Univ. of Sci. & Technol., Tehran, Iran)

Squeak and rattle (S&R) noise are important in-cabin sources of annoyance for occupants. Originally, S&R is generated as a result of colliding and slamming of car's trim and body structure, which in turn occurs because of the dynamic displacements of components excited by road and powertrain. Squeak noise is interpreted as periodic stick and slip phenomena in the contact boundary of two neighbor surfaces. While rattle noise is the emitted noise when adjacent surfaces collide and impact. Both squeak and rattle phenomena happen in high-frequency range, even though the excitation sources (road and powertrain line) works in low frequency range. In practice, squeak and rattle noise can be minimized via controlling the gaps through a tolerance analysis, as well as the appropriate choice of materials. In this research, potential S&R sources are investigated for the trimmed door of a car using clearance analysis. Impact statistics and overall force level at potential rattle places are calculated through random vibration excitation analyses and afterwards, acoustic sensitivity, overall acoustic response, and loudness are calculated by the aim of hybrid SEA-FE method. The results of this prediction will be used in noise and vibration control plan of the whole car in design phase.

2:20

3pSAb5. Intrinsic characterization of structure-borne sound sources and isolators from *in-situ* measurements. Andy Moorhouse, Andy Elliott (Acoust. Res. Ctr., Univ. of Salford, Newton Bldg., Salford M20 1JJ, United Kingdom, a.t.moorhouse@salford.ac.uk), and Yong Hwa Heo (Ctr. for Noise and Vib. Control, KAIST, Daejeon, South Korea)

The paper addresses the problem of how to characterize vibration sources and isolators with measurements made *in-situ*, either on a working installation or on a test bench. For example, automotive components are often characterized by test bench measurements, but there is a need to know how they will behave when coupled to components with potentially different properties in a vehicle. Ideally all components should be characterized by intrinsic properties, which can then be transferred to other installations. In the paper, some novel *in-situ* measurement methods for obtaining these properties are presented. First, the active properties of a source are characterized by the blocked force measured *in situ*. Structural dynamic properties are represented by frequency response functions (mobilities) and it is shown how, if necessary, these may be obtained by indirect measurements, for example when access to measurement points is difficult. New results for dynamic stiffness of isolators are then presented obtained using a novel *in-situ* measurement approach. The method allows rotational (moment) as well as translational (force) dynamic stiffness to be obtained over a wider frequency range than many test rigs. Results are validated by measurement on an ideal laboratory structure.

2:40

3pSAb6. A320 flight deck shape effect on turbulent boundary layer auto-spectrum. Olivier Collery, Manuel Etchessahar, and Miloud Alaoui (Acoust. and Environment Dept., AIRBUS OPERATIONS SAS, 316 route de Bayonne, Toulouse 31060, France, olivier.collery@airbus.com)

The turbulent boundary layer excitation is one of the main sources of aircraft interior noise over a large frequency range and in particular in the flight deck where its geometry drives the physics of the turbulences. The present study investigates measured turbulent boundary layer auto-spectrum properties. For that purpose an A320 has been instrumented with 15 flush-mounted microphones on the upper right quarter of the flight deck. The flight test campaign has been performed in cruise conditions between 27,000 and 39,000 ft with Mach numbers from 0.7 up to 0.82. Analysis of measured data shows strong shape effect on the aerodynamic excitation near windows area. This study points out that advanced numerical tools are required to model these complex aerodynamic phenomena.

Session 3pUW

Underwater Acoustics and Signal Processing in Acoustics: Underwater Acoustic Communications

Mohsen Badiey, Chair
School of Marine Sci. and Policy, Univ. of Delaware, 114 Robinson Hall, Newark, DE 19716

Contributed Papers

1:00

3pUW1. Multiband transmissions for underwater acoustic communication. Aijun Song and Mohsen Badiey (School of Marine Sci. and Policy, Univ. of Delaware, 114 Robinson Hall, Newark, DE 19716, ajsong@udel.edu)

Underwater acoustic communication is important to a variety of scientific and commercial missions in the ocean, for example ocean exploration and monitoring. Due to hardware limitations, often limited bandwidth, for example, 6–7 kHz, has been used in underwater communication systems. In addition to high spectral efficiency, large bandwidth can also lead to increased data rates. When utilizing a large frequency band, frequency-division multiplexing, which refers to dividing an available frequency band into smaller sub-bands, is the common practice. We propose to use multiband transmissions for the underwater acoustic channel, where the wide frequency band is divided into multiple separated sub-bands. The sub-band is much wider than the sub-carrier in the orthogonal frequency-division multiplexing (OFDM). The former is several kilohertz in width while the latter is often only tens of hertz. During our experiment in Hawaii in 2011, high data rates were achieved through the use of multiband transmissions, combined with time reversal demodulation. In the meeting, we will present the receiver algorithms for single- and multi-source acoustic communication systems in the multiband transmission framework. Comparison between the multiband transmissions and OFDM schemes will be also discussed. [Work supported by ONR Code 322OA.]

1:20

3pUW2. Application of differential amplitude and phase-shift keying in underwater acoustic communication based on orthogonal frequency division multiplexing. Pan Zhengrong, Wang Chi, Han Xiao (Sci. and Technol. on Underwater Acoust. Lab., Harbin, Heilongjiang, China), and Yin Jingwei (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China, yinjingwei@hrbeu.edu.cn)

With the increase of of marine resource and underwater users, developing a high-bit-rate underwater acoustic communication has become a hot topic. Amplitude and phase-shift keying (APSK) is a modulation technique having high efficiency in spectrum utilization, it attracts more and more attention in high-bit-rate underwater acoustic communication. Differential APSK (DAPSK) is modulated using differential amplitude and phase code in time domain, and it has higher bandwidth efficiency and be easier to realize the system than APSK. A transmission system based on DAPSK modulation and OFDM is presented to solve the problems of effectiveness and reliability in high-bit-rate underwater acoustic communication. Simulation results show that the system using DAPSK modulation has a better performance than those using APSK and QAM modulation.

1:40

3pUW3. Study on Doppler effects estimate in underwater acoustic communication. Zhang Xiao, Han Xiao (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China), Yin Jingwei (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China, yinjingwei@hrbeu.edu.cn), and Sheng Xueli (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China)

Two estimate methods of the Doppler effects in mobile underwater acoustic communication have been proposed. For the first method the Doppler coefficients are obtained by estimating frequency change of CW impulse signal with notch filter. For the other method the Doppler coefficients are obtained by estimating chirp rate change of LFM signal with fractional Fourier transform (FRFT). And the performance of the Doppler effects estimation method based on notch filter or the FRFT are compared by the computer simulation. The advantages, shortcoming and the application occasion of the two methods are elaborated. The effectiveness and robustness of the two methods have been proved. Key Word: Notch Filter, Fractional Fourier transform, Doppler Effects, Underwater Acoustic Communication

2:00

3pUW4. Influence of multipaths on coherent acoustic communication in shallow water channel. Su-Uk Son and Jee Woong Choi (Dept. of Marine Sci. and Convergence Technol., Hanyang Univ., 55 Hanyangdaehak-ro, Sangnok-gu, Ansan, Gyeonggi-do 426-791, South Korea, suuk2@hanyang.ac.kr)

In shallow water communication channel, acoustic interactions with sea surface and bottom interfaces cause the inter-symbol interference that hinders the efficient and reliable communication. In this case, signal-to-multipath ratio (SMR) rather than signal-to-noise ratio can be used as an indicator to describe the quality of the communication channel. However, it is difficult to estimate precisely the SMR from the measured communication data. In this talk, we propose the energy fraction of the channel impulse response existing within one symbol duration as an alternative to SMR. Communication experiment was conducted on the southern coast of Korea in waters 45 m deep in source-receiver ranges of 100 m to 1 km. The bit-error-rate performance is compared to the energy fraction in one symbol duration. In addition, the correlation between the energy fraction in a symbol and SMR is investigated through a Monte Carlo simulation. [Work supported by ADD (Agency for Defense Development, Korea).]

2:20

3pUW5. Study on time reverse mirror in underwater acoustic communication. Yin Jingwei (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China, yinjingwei@hrbeu.edu.cn), Du Pengyu (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China), Shen Jianwen (Kunming Shipbuilding Electron. Equipment Co. Ltd, Kunming, China), and Guo Longxiang (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China)

Time reversal mirror (TRM) can adaptively match the sound channel without any prior knowledge. In this paper, active TRM, passive TRM and virtual TRM which are all based on a single array element and the application of TRM in the underwater acoustic communication including single-user communication and multi-user communication are studied. Single sensor TRM which has time compression performance lacks array processing space gain, however, it can meet the requirements for underwater acoustic communication nodes being in the pursuit of simple structure and low power consumption. It is verified that TRM could focus multipath signal and achieve real-time adaptive channel equalization through computer simulation and test results, which could suppress the inter-symbol interference(ISI) and improve the signal-to-noise ratio(SNR).

2:40

3pUW6. Study of underwater speech coding technique based on contact conduction transmitter. Xuelli Sheng, Ye Bai, Jia Lu, Jin Han, and Weijia Dong (Harbin Eng. Univ., Shuisheng Bldg., 803#, Harbin, China, sheng-xueli@yahoo.com.cn)

Low bit rate speech coding of 2.4 kbps is actualized by mixed excitation linear prediction algorithm, which meets the requirement of high data rate of underwater speech communication system and the limit of high communication rate of underwater acoustic communication technique. In the process of speech coding, a great deal of speech data is compressed to high speed underwater acoustic communication, and the main characteristics of speaker are remained perfectly. Meanwhile, the capability of avoiding interference is improved obviously by the speech based on contact conduction transmitter as input signal. This underwater speech coding technique and differential OFDM technique are combined and experimented under 434, 1310, and 2000 m in the lake. The real-time transmission of speech signal is presented in the condition of complicated multi-path in the extremely shallow sea. The results show the synthesized speech has well quality of intelligibility and clarity, which satisfies the demand of underwater speech coding.

Plenary Session and Awards Ceremony

David L. Bradley
President, Acoustical Society of America

Christian Giguère
President, Canadian Acoustical Association

Michael Vorländer
President, International Commission for Acoustics

Acoustical Society of America

Presentation of ASA Fellowship Certificates

Peter F. Assmann – For contributions to vowel perception and the influence of talker variability on speech patterns
Li Cheng – For contributions to vibroacoustic modeling of complex structures
M. Patrick Feeney – For contributions to clinical middle-ear function through wideband reflectance
Eric W. Healy – For contributions of spectral-temporal analysis in speech perception
Philip X. Joris – For contributions to neural encoding in binaural hearing
Michael V. Scanlon – For contributions to the development of systems to detect and localize transient sounds in air
Michael Verslusis – For contributions to high speed imaging of fine scale acoustic phenomena

Presentation of Acoustical Society of America Awards

William and Christine Hartmann Prize in Auditory Neuroscience to Tom C. T. Yin
Medwin Prize in Acoustical Oceanography to Philippe Roux
R. Bruce Lindsay Award to Eleanor P. J. Stride
von Bèkésy Medal to M. Charles Liberman
Helmholtz-Rayleigh Interdisciplinary Silver Medal to Timothy J. Leighton
Gold Medal to Lawrence A. Crum

Canadian Acoustical Association

Announcement of Canadian Acoustical Association Award Recipients

International Commission for Acoustics

Presentation of the ICA Early Career Award to Tapio Lokki

Session 3eED

Education in Acoustics: Women in Acoustics—Listen Up and Get Involved

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

Marcia J. Isakson, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713

This workshop for Montreal area Pathfinder Girl Guides (ages 12–18) consists of a hands-on tutorial, interactive demonstrations, and a panel discussion about careers in acoustics. The primary goals of this workshop are to expose the girls to opportunities in science and engineering and to interact with professionals in many areas of acoustics. A large number of volunteers are needed to make this a success. Please email Traci Neilsen (tbn@byu.ed) if you hav time to help with either guiding the girls through the tutorial led by Wendy Adams (5:00 p.m.–6:15 p.m.) or exploring principles and applications of acoustics with small groups of girls (5:45 p.m.–7:30 p.m.). We will provide many demonstrations, but feel free to contact us if you would like to bring your own.

WEDNESDAY EVENING, 5 JUNE 2013 7:00 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees on the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings.

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 Wednesday are as follows:

7:30 p.m. Biomedical Acoustics	519a
7:00 p.m. Signal Processing in Acoustics	510a