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
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

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: $T_{sub} = 1.41$ 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
Psych., 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—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.]

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."

3a WED. AM

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

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 multi-component 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
Dept. of Human Sci., Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan

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
Biomedical Eng., Dalhousie Univ., 1276 South Park St., DCson Bldg. rm 3191, Halifax, NS B3H2Y9, Canada

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 spatiotemporal 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

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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.