Session 5aAAa

Architectural Acoustics and Psychological and Physiological Acoustics: Psychoacoustics in Rooms II

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Chair's Introduction—8:00

Invited Papers

8:05

5aAAa1. Some effects of room acoustics and background noise on assistive listening systems and devices. Peter Mapp (Peter Mapp Assoc., Copford, Colchester CO6 1LG, United Kingdom, petermapp@petermapp.com)

Approximately 10–14% of the general population (United States and Northern Europe) suffer from a noticeable degree of hearing loss and would benefit from some form of hearing assistance or deaf aid. However, many hearing aids do not provide a suitable level of intelligibility when used in large reverberant or noisy spaces and rooms. The paper investigates the acoustic and speech intelligibility requirements for Assistive Listening Systems that may be used separately or in conjunction with a hearing aid to improve the potential intelligibility of the received speech signal. A number of microphone pick-up scenarios have been investigated and are reported in terms of their potential intelligibility and sound quality performance. The results of testing carried out in a number of rooms and venues are presented, mainly in terms of the resultant Speech Transmission Index (STI). The paper concludes by providing a number of recommendations and guidelines for successful microphone placement and introduces a novel technique for establishing useful talker sound radiation and hence microphone target aiming.

8:25

5aAAa2. Investigation of subjective components of overall acoustic quality using binaural recordings made in Hartford's Belding Theater. Acadia A. Kocher (Northwestern Univ., 6120 Holly Ridge Ct, Columbia, Maryland 21044, acadiakocher2015@u.northwestern.edu) and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

Previous research in concert hall acoustics has established correlations between listener preference for overall acoustic quality (OAQ) and a number of other subjective factors. A subjective study was conducted to determine the relative importance of several characteristics to OAQ. Subjects evaluated 18 signals based on listener envelopment, reverberance, tonal quality, and overall preference using five-point rating scales. The signals were presented individually over headphones. A total of 32 subjects with formal musical training and normal hearing were included in the study. Following a brief tutorial and training session, the test was divided into four sets, one for each of the subjective factors. All of the signals and sets were presented to each subject in a random order. The binaural recordings were made in the Belding Theater in Hartford, Connecticut. A loudspeaker on the stage played three short classical motifs that were recorded in three seat locations with two distinct settings of the hall's variable acoustics system. The sound pressure levels of all signals were normalized to isolate the characteristics of interest. The data analysis identified relationships between the three factors and OAQ, with the most significant relationship between OAQ and tonal quality. [Work was supported by NSF Grant 1302741.]

8:45

5aAAa3. Localizing low-frequency sounds in a room and the dominance of interaural level differences. William M. Hartmann, Brad Rakerd, Eric J. Macaulay, and Zane D. Crawford (Michigan State Univ., 567 Wilson Rd., East Lansing, MI 48824, hartman2@msu.edu)

In a room-environment study of the duplex theory of sound localization, listeners reported the azimuthal locations of low-frequency sine tones in free field and in three very different rooms. Interaural differences in time (ITD) and level (ILD) were continuously monitored by probe microphones in the listeners' ear canals. As the frequency increased from 250 to 1000 Hz, the correlation of listener responses with the ILD increased while the correlation with the ITD decreased precipitously. The increased importance of the ILD was especially prominent for the least reverberant environments. The decreased importance of the ITD occurred primarily because of interaural phase differences (IPD) that became large (>90 degrees) and consequently weak. The large IPD effect occurred more frequently in the highly reverberant room, and in the less reverberant rooms at 750 and 1000 Hz, where the peak of the IPD distribution occurred well above 90 degrees. The two effects caused the ILD to become more important than the ITD when the frequency was greater than about 500 Hz with only a small dependence on the different rooms. The increased emphasis given to the ILD normally led to more accurate localization. [Work supported by AFOSR grant FA9550-11-1-0101.]

9:05

5aAAa4. On the sensitivity of older hearing-impaired individuals to acoustic attributes. William M. Whitmer and Michael A. Akeroyd (Scottish Section, MRC/CSO Inst. of Hearing Res., Glasgow Royal Infirmary, Glasgow, United Kingdom, bill@ihr.gla.ac.uk)

In previous work, we have run a series of experiments with hearing-impaired adults to examine how age and hearing loss affect sensitivity to changes in apparent auditory source width. In two experiments, the interaural coherence of broadband noises presented over headphones and loudspeakers was varied to induce changes in width; in a third, older participants sketched the width of noise, speech, and musical stimuli in simulated rooms with varying reflection absorption. The results of those experiments showed generally decreasing sensitivities to interaural-coherence-induced changes in width as a function of age and hearing impairment. To examine how these results might influence acoustic design for the aged, a new study considers a more basic task that avoided the auditory-visual transformation implicit in sketching. Participants with normal to moderately impaired hearing will compare simulated utterances in a same/different room-discrimination task. Binaural impulse responses will be generated for open-plan buildings of varying size from the ODEON database with sources and receivers centered in each space, convolved with speech and music tokens and presented over headphones. The differences in the relationship between presbycusis and sensitivity to source width vs. general acoustical attributes will be discussed. [Work supported by the MRC and the CSO.]

9:25

5aAAa5. Amplitude modulation sensitivity in rooms. Pavel Zahorik (Heuser Hearing Inst. and Div. of Communicative Disord., Dept. of Surgery, Univ. of Louisville School of Medicine, Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu)

The physical effects of room acoustics on sound emitted from an amplitude-modulated (AM) source is well understood and can be effectively characterized by the modulation transfer function (MTF) of the room. Human sensitivity to AM is also well understood under conditions which minimize the acoustical contributions of the listening environment, and can analogously be characterized using the MTF concept. Although it may be predicted that AM sensitivity in a reverberant soundfield could be explained if both AM sensitivity in the absence of the soundfield and the MTF of the room are known, little empirical data exist in the literature to validate this prediction. Here, we summarize recent work in our laboratory that addresses this issue. A consistent finding is that predicted AM sensitivity underestimates measured sensitivity for reverberant soundfield listening. This result appears to depend critically on prior listening exposure to the soundfield, and is consistent with recent neural data from other laboratories. It may also explain why speech understanding in normal-hearing populations is typically unaffected by the acoustical degradations caused by reverberation. [Work supported by NIH-NIDCD.]

9:45

5aAAa6. Meeting the classroom acoustics standard on a historical room. Ana M. Jaramillo (Olson Sound Design, 3711 Lake Dr., 55422, Robbinsdale, MN 55422, ana.jaramillo@afmg.eu) and Bruce C. Olson (Olson Sound Design, Brooklyn Park, MN)

The Department of Speech-Language-Hearing Sciences at University of Minnesota occupies Shevlin Hall built in 1906. We were called in 2013 to help with the acoustic and sound system redesign of room 110 to be used as a classroom, as the room had very poor speech intelligibility due to the very high ceiling and hard walls, resulting in a very reverberant environment. Our goal was to meet the current classroom acoustics standard ANSI S12.60. Reverberation time and noise measurements were performed, and the room was modeled using EASE for the prediction of results. The model was compared with measurements, and several alternatives were explored. After the changes were implemented, measurements confirmed the improvement to the room's intelligibility was acquired and the historic character of the room was preserved.

10:05–10:15 Break

Contributed Papers

10:15

5aAAa7. Perception of loudness in directional sound fields. Philip W. Robinson (Media Technol., Aalto Univ., PL 15500, Aalto 00076, Finland, philrob22@gmail.com) and Jukka Pätynen (Media Technol., Aalto Univ., Espoo, Finland)

An often utilized assumption in room acoustics is that the room produces a diffuse field, in which sound is uniformly distributed in space and arrives at the listener equally from all directions. This assumption greatly simplifies many calculations, e.g., Sabine's reverberation time (RT). However, the reverberant field, particularly the early part of the impulse response in typical concert spaces, is highly directional. Hence, the diffuse assumption leads to errors, for example, in the prediction of loudness when using omni-directional measures. Lateral reflections contribute to perceived loudness more than their omni-directionally measured sound energy. This is due the filtering of the head and torso, which amplify reflections from lateral directions more than others, particularly at high frequencies. Listening test results will be presented that demonstrate this effect. Spatial analyses of concert hall impulse responses demonstrate the practical applicability of this finding and make evident the relevance of the effect.

10:30

5aAAa8. Temporal weighting of binaural cues in real rooms: Psychophysical and neural modeling. G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu), Andrew D. Brown, and Daniel J. Tollin (Dept. of Physiol. and Biophys., Univ. of Colorado School of Medicine, Denver, CO)

Although interaural time and level differences (ITD and ILD) comprise the primary cues to sound location in azimuth, both are distorted by echoes and reverberation in many real environments. Consequently, the "effective" cues exhibit complexities and evolve temporally due to interactions of direct and reflected sound. One approach to studying the perceptual effects of such distortions is to measure the relative influence, or "temporal weight," of cues contained within brief temporal segments of sound on spatial judgments made by human listeners. Temporal weighting functions (TWFs) measured in this way reveal binaural sensitivity to be temporally nonuniform and cue-specific. For many sounds (tones and high-rate click trains), judgments appear dominated by the ITD and ILD occurring at sound onset and the ILD occurring near sound offset, while middle portions contribute very little, consistent with expectations about the temporal statistics of these

cues in real rooms. In this presentation, psychophysically derived TWFs are compared to the frequency-dependent statistics of ITD and ILD in room recordings. Models of the auditory periphery and early nervous system are used to transform the recordings to estimate the effective cues "as heard by" central brain mechanisms. [Work supported by NIH R01 DC 011548 (GCS) and DC 011555 (DJT).]

10:45

5aAAa9. Designing an auditory lab including active acoustics. Ranil Sonnadara (McMaster Inst. for Music and the Mind, McMaster Univ., Hamilton, ON, Canada), Steve Ellison (Meyer Sound Labs, Inc., 2832 San Pablo Ave., Berkeley, CA 94702, ellison@meyersound.com), Laurel Trainor, and Dan Bosnyak (McMaster Inst. for Music and the Mind, McMaster Univ., Hamilton, ON, Canada)

LIVELab, a new facility at McMaster University, has been developed in order to help support research in areas such as developing intelligent hearing aids, measuring the physiological impact of media presentations, understanding interactions between musicians, and determining effective sounds for medical and warning systems. This room combines a quiet performance space featuring a low nominal reverberation time with technology including active acoustics, multichannel playback, and integral measurement and instrumentation to create a sonically flexible facility for auditory research. The design of the room and its acoustic variability for supporting these research endeavors will be discussed.

11:00

5aAAa10. Perceptually evaluating ambisonic reproduction of room acoustics. Samuel Clapp (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, clapps@rpi.edu), Anne Guthrie (Arup Acoust., New York, NY), Jonas Braasch, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Spherical microphone array technology allows for the recording of auditory scenes in three dimensions by decomposing a soundfield into its spherical harmonic components. The performance of the array is affected by certain factors in its design, including the size of the array, whether an open or a rigid sphere configuration is used, and the number of sensors and their placement scheme. Ambisonics is a system designed to reconstruct a soundfield from its spherical harmonic components. The process of ambisonic decoding determines the signals that are fed to each loudspeaker in the array to simulate a given soundfield. Such systems have certain accuracy constraints, particularly at higher frequencies, and different decoding methods can be used at those frequencies to recreate more accurate spatial cues, particularly ILD cues. This paper examines how to develop a decoding scheme that addresses the constraints in both the recording and playback phases, and uses binaural modeling to determine its efficacy. Reconstruction of both simulated and real rooms is examined, with respect to the accurate reproduction of important room acoustic metrics.

11:15

5aAAa11. Subjective perception of varying reflection densities in room impulse responses. Hyun Hong and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, hhong@huskers.unl.edu)

The commonly used objective metrics for analyzing room acoustics are each tied to some aspect of subjective perception. For example, the widely used reverberation time is linked to the perceived reverberation in a room. Two different rooms having the same reverberation time, though, can have different reflection densities in their impulse responses, and this difference in reflection density may affect how listeners perceive the size of the rooms. This project investigates the subjective perception of reflection densities and how sensitive humans are to a change of reflection density. First, assorted parameters for quantifying reflection density are reviewed. Then, preliminary results of perceptual tests using different room impulse responses with varying reflection densities are presented.

11:30

5aAAa12. Influence of manipulated early reflections on room acoustical perception. Stefan Klockgether and Steven van de Par (Appl. Phys. and Acoust., Carl von Ossietzky Univ., Carl-von-Ossietzky-Str. 9-11, Oldenburg, Lower Saxony 26129, Germany, stefan.klockgether@uni-oldenburg.de)

The binaural room impulse response (BRIR) can be used to study the perception of room acoustics and consists of different parts such as the direct sound, early and late reflections, and a diffuse reverberant tail. For this study, the contribution of early reflections to several perceptual attributes was investigated. For this purpose, the strength of the early reflections of recorded BRIRs were either increased or decreased. The resulting BRIRs as well as the original BRIRs have then been convolved with anechoic music signals to obtain the stimuli that were presented to the subjects in a psychoacoustic experiment. The subjects had to rate the spatial impression of these manipulated and none-manipulated signals for the perceptual attributes "Listener envelopment,"" Apparent source width," "Distance" and "Presence." In addition to determining the influence of manipulated early reflections on the spatial impression of a room, also a comparison was made with the perceptual effect of an artificial increase of the interaural cross-correlation. Results indicate that the perceived source width increases with increasing level of the early reflections. The effect of the level manipulations on the listener envelopment seems to be small compared to the influence of the cross-correlation.

Session 5aAAb

Architectural Acoustics: Exploring the 2014 Sound and Vibration Guidelines and Case Studies for Healthcare Facilities

Kenric D. Van Wyk, Cochair
Acoustics By Design, Inc., 124 Fulton St. East, Second Fl., Grand Rapids, MI 49503

Daniel M. Horan, Cochair
Cavanaugh Tocci Associates, Inc., 327 F Boston Post Rd., Sudbury, MA 01776

Edward Logsdon, Cochair
D. L. Adams Associates, Inc., 1536 Ogden St., Denver, CO 80218

Chair's Introduction—8:00

Invited Papers

8:05

5aAAb1. Changes in the acoustical requirements of the 2014 editions of the Facility Guidelines Institute's *Guidelines for Design and Construction of Hospitals and Outpatient Facilities* and its reference document *Sound & Vibration Design Guidelines for Health Care Facilities*. Daniel M. Horan (Cavanaugh Tocci Assoc., Inc., 327 F Boston Post Rd., Sudbury, MA 01776, dhoran@cavtocci.com)

Acoustical design criteria for health care facilities as defined by the FGI *Guidelines* have been updated in the recently published 2014 edition. *Sound & Vibration Design Guidelines for Health Care Facilities* (S&V3.0) serves as the sole acoustical reference for the FGI *Guidelines* and has also been updated as part of the 2014 FGI cycle. The S&V reference is authored and edited by the FGI's acoustical working group. The secretary of this group (Horan) will summarize these changes and will also discuss the public review and comment process that led to the updated *Guidelines*. This same process will be used in the forthcoming cycle in preparation for the 2018 edition(s).

8:25

5aAAb2. Evidence based design for improved patient experience. Melinda Miller (Acoustics By Design, Grand Rapids, MI), Kenric Van Wyk, and Kristen Murphy (Acoustics By Design, 124 Fulton St. E, Grand Rapids, MI 49503, kvanwyk@acousticsbydesign.com)

In addition to the health benefits of improving acoustical comfort for patients and staff in healthcare environments, there is now a financial incentive. Since October 2012, Value Based Purchasing reimbursements of Medicare from the Federal Government is dependent in part upon Hospital Consumer Assessment of Healthcare Providers and Systems (HCAHPS) patient satisfaction survey results, which includes a rating for the noise level at night. This study explores the relationship of sound level with their associated patient satisfaction scores from hospital noise surveys performed by Acoustics By Design. The noise study results are presented across different parameters such as the type of patient unit, patient noise levels, nurses' station noise levels, time of day, and private versus semi-private rooms. Measurements were also performed prior to and after completion of acoustical/noise mitigation.

8:45

5aAAb3. Modular construction in healthcare. Edward Logsdon and Benjamin Seep (Acoustical Consulting, D. L. Adams Assoc., Inc., 1536 Ogden St., Denver, CO 80218, elogsdon@dlaa.com)

Healthcare facilities are utilizing modular constructions where elements like the patient restroom or headwall are built off-site and then shipped to the project where they are then installed. The modular assemblies are constructed under tight quality control and the manufacturers suggest savings in labor, shortened schedules, and improved architectural and acoustical performance. Modular patient restroom assemblies were used in the construction of a new major Colorado hospital with 360 private patient rooms. This includes the restrooms used in the labor and delivery areas of the hospital. Acoustical plus and minuses with the use of the modular patient restrooms will be discussed along with recommendations to maintain sound isolation between the spaces. This includes detailing during design and modifications needed to address issues in the field.

9:05

5aAAb4. The implication of door undercut in patient room to corridor speech privacy. Gregory C. Tocci (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA 01776, gtocci@cavtocci.com)

Un-gasketted corridor door undercuts and un-gasketted head, hinge, and lockset jambs compromise the STCc rating of corridor wall/door assemblies. However, health care infection control policies often do not permit gasket systems on many doors. Though loss of sound isolation is obvious, this paper specifically characterizes the loss of speech privacy between patient rooms and corridors with doors not provided with undercut drop seals and jamb gaskets. The work draws upon that of Kim and An, and characterizes speech privacy conditions using the articulation index (AI) and its compliment the privacy index (PI).

9:25

5aAAb5. The effect of implementing an electronic sound masking system into a 42-bed oncology unit on "quiet at night" Hospital Consumer Assessment of Healthcare Providers and Systems (HCAHPS) scores. Gary Madaras (Acoust., Making Hospitals Quiet | Rockfon, 4849 S. Austin Ave., Chicago, IL 60638, DoctorSonics@aol.com)

In December 2012, the facilities personnel at a major New York medical center used a donation to install an electronic sound masking system in the patient rooms of a 42-bed oncology care unit. Monthly top-box scores for the unit's 'quiet at night' HCAHPS question as reported by Press Ganey for the years preceding and succeeding the installation will be provided and discussed. Other observations based on staff interviews, personal listening, and acoustical measurements will be provided.

9:45–10:00 Break

10:00

5aAAb6. Integrated design experience and post-occupancy acoustic performance of medical office building of the future. Erik Miller-Klein (SSA Acoust., LLP, 222 Etruria St, Ste. 100, Seattle, WA 98109, erik@ssaacoustics.com)

This case study discusses the experiences working within this integrated design team with architects, engineers, and contractors from day one, which included mock-ups and physical models of proposed systems. This project was completed in 2013, and was designed and evaluated compared to the 2010 Facilities Guidelines Institute Guidelines for Design and Construction of Health Care Facilities. During construction, we evaluated some acoustic construction defects and addressed these impacts, and completed a full acoustic evaluation of the space prior to occupancy. This building was a paradigm shift for the doctors and nurses, and required additional testing and analysis to interpret some of the feedback from users about the acoustic function of the spaces.

10:20

5aAAb7. An approach for estimating noise induced hearing loss because of indoor and outdoor environment noise factors in healthcare facilities. Filiz B. Kocyigit (Architecture, Atilim Univ., Kizilcasar Mah. Incek, Ankara 06560, Turkey, filizbk@gmail.com)

This paper investigates a number of healthcare centers from University Medical Schools and State Hospitals, Baskent University Medical School and Private Research Hospitals, and the State Hospital. The sound levels present in these centers were compared with number international as well as Turkish standards. Research was carried out in the following way: First, physicians and nurses were surveyed by means of a questionnaire. Then, the hospital buildings were examined in terms of their architectural design features. Lastly, sound levels were measured in the consequent spaces where questionnaires were also conducted. In addition, noise in hospitals can be detrimental as it helps to present their environment as quiet and peaceful. WHO provides guidelines for hospitals in this respect in its Guidelines for Community Noise published in 1995 (4). ANSI S12.2, recommends a maximum Room Criterions Curves (NCC) value ranging on the room type, and a maximum Noise Criteria Balanced (NCB) value ranging. A document issued by the Environmental Protection Agency (EPA) summarizing significant community noise studies provides recommendations in terms of the Ld&n (day-night sound pressure level). The Institute of Turkish Standards (TSE) is also working on this subject and the first national standard about ambient at Table 11.

10:40

5aAAb8. A review of hospital noise studies. Melinda Miller, Kenric Van Wyk, and Kristen Murphy (Acoustics By Design, 124 Fulton St. E, Grand Rapids, MI 49503, melinda@acousticsbydesign.com)

Hospital noise concerns are on the rise due to detrimental health effects and the implementation of Value Based Purchasing, which affects federal reimbursements based on HCAHPS patient satisfaction scores. Because noise is one of the lowest performing categories in the survey, it is expected that demand for hospital noise surveys will increase. Therefore, it is useful to examine existing hospital noise measurement techniques to serve as a guideline for future work. This paper provides a summary of several noise studies of existing healthcare facilities conducted by Acoustics By Design (ABD) and others. The goal of the studies performed by ABD was to balance providing a comprehensive noise study (across multiple noise variation factors, such as time of day, and weekend and weekday conditions, department type, etc.) with providing a study that is time and cost efficient. We have found that this is best done through a combination of short term and long term measurements in close collaboration with hospital staff and administration.

Contributed Papers

11:00

5aAAb9. Trends of the acoustic condition in an intensive care unit based on a long-term measurement. Munhum Park (Smart Sensing & Anal., Philips Res. Labs., High Tech Campus 36.p.078, Eindhoven 5656AE, Netherlands, mun.park@philips.com), Piet Vos (Dept. of Intensive Care, St. Elisabeth Hospital, Tilburg, Netherlands), Armin Kohlrausch (Smart Sensing & Anal., Philips Res. Labs., Eindhoven, Netherlands), and Annemarie W. Oldenbeuving (Dept. of Intensive Care, St. Elisabeth Hospital, Tilburg, Netherlands)

Noise levels in hospitals, especially in intensive care units (ICUs), are often very high, potentially influencing the patients' well-being and recovery processes, where the undesirable acoustic environment is also considered to be one of the risk factors contributing to ICU delirium. In the current study, a continuous measurement was taken for 3 months in 8 single-bed patient rooms in an ICU, of which the results were analyzed in synchrony with the admission of 106 patients. On average, the A-weighted energy-equivalent sound pressure level (L_{Aeq}) in patient rooms varied significantly with the time of day ($p < 0.001$), but was not dependent on the day of week ($p = 0.448$). Furthermore, analysis of noise levels in occupied versus unoccupied rooms indicated the dominance of room-internal sources in the former and room-external sources in the latter periods. During the first four days of patients' ICU stay, the acoustic condition improved slightly from day 1 to day 2, but the noise level rebounded from day 2, most likely in relation to the various phases of treatment and recovery. Between-patient variability was found to be significant, which may be an important aspect to consider when comparing the acoustic conditions in an environment occupied by different groups of patients.

11:15

5aAAb10. Study to optimize speech clarity in a hospital pediatric trauma room, using newly patented tuning tube and a custom nonwoven fabric. Bonnie Schnitta (SoundSense, LLC, 46 Newtown Ln., Ste. One, East Hampton, NY 11937, bonnie@soundsense.com)

There are several acoustic requirements in order to optimize the healing environment for the patients in a trauma room, as well as increase medical staff concentration and clarity of speech within the room. This is especially true in a pediatric setting. First, there is the requirement that the reverberation time of the room must be reduced in order to assist in hearing/speech clarity while adhering to clean room standards. Secondly, there needs to be an increase in the signal to noise ratio (SNR). A greater SNR assists in hearing/speech clarity; the lower reverberation time also assists in preventing the various intrusive mechanical sounds from becoming amplified and further reducing the SNR. In addition to these two standard room requirements, there is also a need for the reduction of lower frequency sounds within a room, such as sounds typical of mechanical equipment. Since standard products used to absorb sound have more absorption in speech frequencies, there is a need for products that have greater absorption in lower frequencies. This paper presents background on two SoundSense patented products that were used in a study for improved conditions and outcomes in a hospital pediatric trauma room.

11:30–12:00 Panel Discussion

FRIDAY MORNING, 9 MAY 2014 554 A/B, 8:55 A.M. TO 11:00 A.M.

Session 5aAB

Animal Bioacoustics and Education in Acoustics: Communicating the Science of Underwater Sound

Kathleen J. Vigness-Raposa, Chair
Marine Acoustics, Inc., 809 Aquidneck Ave., Middletown, RI 02842

Chair's Introduction—8:55

Invited Papers

9:00

5aAB1. The science of underwater sound: Education, communication, and outreach. Gail Scowcroft (Graduate School of Oceanogr., Univ. of Rhode Island, 1 Ferry Rd., Narragansett, RI 02882, gailscow@mail.uri.edu) and Kathleen J. Vigness-Raposa (Marine Acoust., Inc., Middletown, RI)

As a complex scientific topic, underwater sound can be challenging for scientists to discuss and effectively communicate with non-science audiences. Educational audiences span formal K-16 classrooms to museum and aquarium visitors. The science of sound is often included in upper middle school physical sciences curricula, high school physics classes, and undergraduate and graduate university courses, which can take advantage of calculus to support student understanding. Communicating with the media presents other challenges: pressing or immediate deadlines; a need to deliver eye-catching, flashy pieces that capture reader attention; and a general lack of fundamental knowledge of underwater sound by readers. Scientists must be proactive in their engagement with media to ensure good

fundamental science is communicated and to increase useful stories about new developments in underwater sound research. Regulators and other decision-makers are also pressed for time when contemplating a topic, yet they need the most up-to-date scientific findings to support their decision-making. This talk will provide an overview of the challenges that ocean acoustic specialists face when trying to communicate the results of their research and meet the needs of diverse audiences. In addition, strategies and possible solutions will be discussed.

9:20

5aAB2. Student engagement and education in underwater sound through the build-a-hydrophone activity. Kevin Joy (Northeast Underwater Res. Technol. and Education Ctr., Univ. of Connecticut, 1080 Shennecossett Rd., Groton, CT 06340, kevin.joy@uconn.edu)

The hydrophone is arguably the most basic sensor used in underwater acoustics. In its simplest form, this passive device provides scientists, and users of all kinds, the ability to listen to and capture natural and anthropogenic sounds that occur in our aquatic world. As part of an array, this technology enables humans to perform such underwater tasks as tracking submerged targets to studying volcanic and tectonic processes. The concept behind the do-it-yourself (DIY) activity to build a hydrophone is not unique. A simple Internet search will reveal numerous design options geared toward constructing a functional hydrophone. The Center for Ocean Sciences Education Excellence (COSEE), Technology and Engineering for Knowledge (TEK), at the University of Connecticut, has modified this activity to offer an alternative that is both affordable and functional, while minimizing the technical skills, tools, and time required for completion. The outcome has been the development of a hands-on student exercise that provides an avenue for engagement and learning in the Science, Technology, Engineering and Mathematics (STEM) fields, while offering a simple means to introduce the science of underwater sound.

9:40

5aAB3. Animations for communicating the science of underwater sound. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu)

For several years, the author has been developing a popular website (http://www.acs.psu.edu/drussell/demos.html) consisting of a variety of computer animations illustrating acoustic phenomena involving wave propagation and vibration [Russell, J. Acoust. Soc. Am., **114**, 2308 (2003)]. This paper will showcase a subset of those animations, including several new animations of acoustic wave phenomena and interactive plots created using the Wolfram Mathematica Computable Document Format, which are specifically geared toward enhancing student understanding of the propagation and radiation of sound underwater. Animation examples will include wave propagation; sound radiation from simple sources; directivity patterns for dipoles, doublets, line arrays, and the baffled piston; refraction and reflection; absorption of sound; and waveguides.

10:00

5aAB4. Able sea chicks... Adventures in acoustical oceanography. Lora J. Van Uffelen (Ocean and Resources Eng., Univ. of Hawaii at Manoa, 1000 Pope Rd., MSB 205, Honolulu, HI 96815, loravu@hawaii.edu) and Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., Fairfax, VA)

The Able Sea Chicks blog introduces readers to the exciting field of acoustical oceanography by following two female scientists on research cruises to deploy acoustic moorings in the Philippine Sea. The blog was designed for a special outreach session at the April 2010 Acoustical Society of America meeting titled “Listen up and get involved,” aimed at girls involved in the Girl Scouts of America. This session included a video explaining some of the principles of underwater sound propagation and showcasing some of the instrumentation used to collect underwater acoustic data. A live, interactive question and answer session with the scientists onboard the research vessel followed the video presentation. Both the blog and the outreach session were designed to reach girls at the age where research has shown that interest in STEM fields begins to decline. The blog illustrates how ocean acoustics experiments are conducted using examples from the PhilSea10 deployment and recovery cruises. Blog posts cover topics such as hydrophone testing, signal processing, mooring deployment, and long-baseline navigation. Since 2010, the Able Sea Chicks video and blog have been used in other Girl Scout sessions at ASA and in talks for high school and university audiences.

10:20

5aAB5. International harmonization of approaches to define underwater noise exposure criteria. Klaus Lucke (CMST, Curtin Univ., GPO Box U1987, Perth, WA 6845, Australia, klaus.lucke@wur.nl), Erwin Winter (Fish Res., IMARES Wageningen UR, IJmuiden, Netherlands), Frans-Peter Lam (Underwater Technol. Dept., TNO, The Hague, Netherlands), Gail Scowcroft (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI), Anthony Hawkins (Loughine Ltd., Aberdeen, United Kingdom), and Arthur N. Popper (Dept. of Biology, Univ. of Maryland, College Park, MD)

An international workshop was held in 2013 with a group comprised of scientists, regulators, and other stakeholders. The workshop focused on how new scientific information related to the effects of underwater noise on marine life influences permitting practices for human activities at sea. Also discussed were how individual countries regulate underwater noise and opportunities for harmonizing approaches on an international scale. The workshop was intended to build momentum toward an international exchange of information and to potentially establish a network for the regulation community. Large gaps in knowledge still exist. In particular, hearing sensitivity in baleen whales, long-term effects of TTS and relevant information on other taxa such as bony fishes, sharks, or invertebrate species, need to be studied more intensively. Regulators need reliable and understandable baseline information on cause-effect relationships. This information could be partially provided through targeted training material for regulators. Another critical regulator need is for opportunities to speak with each other and share knowledge across wide geographic regions. Additional keys to future success are commitments from the regulatory senior management and politicians, invite nations who were not represented in the discussions so far and raise awareness of this topic across a broad audience, including the public.

10:40

5aAB6. Discovery of Sound in the Sea: Resources for regulators, policymakers, and other stakeholders. Kathleen J. Vigness-Raposa (Marine Acoust., Inc., 809 Aquidneck Ave., Middletown, RI 02842, kathleen.vigness@marineacoustics.com), Gail Scowcroft, Christopher Knowlton, and Holly Morin (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI)

There is concern about the effects of underwater sound on marine life. These effects must be considered by users of underwater sound under several regulations. To fulfill regulatory requirements, up-to-date resources are needed on the potential effects of underwater sound, as well as fundamental science content. The Discovery of Sound in the Sea website (DOSITS; http://www.dosits.org) provides accurate scientific information on underwater sound at levels appropriate for all audiences, including the general public, K-12 teachers and students, college students, policy-makers, and professionals in industry, education, and the media. Content, such as the effects of sound on marine life, is based on peer-reviewed publications and has undergone rigorous review by the DOSITS scientific advisory panel. This talk highlights new resources for regulators, including structured tutorials and a "hot topics" feature. Structured tutorials provide a directed progression of sequential knowledge; the first one outlines the scientific process for determining the risk of exposure to underwater sound, answering the question "How do you determine if a sound source might affect a marine animal?" The "hot topics" feature allows for current, topical issues to be highlighted on the DOSITS front page, providing efficient access to new information, as well as foundational scientific content.

FRIDAY MORNING, 9 MAY 2014 557, 8:45 A.M. TO 12:00 NOON

Session 5aNS

Noise: Aircraft and Fan Noise and Analysis

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

Contributed Papers

8:45

5aNS1. Detection and identification of helicopter noise in beach environments. Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

Noise generation by helicopters has been studied extensively as has auditory perception of helicopter noise—both in terms of detection and identification and in terms of annoyance. Recent work [Hoglund *et al*., J. Acoust. Soc. Am. **128**, 164–171 (2010)] has identified the importance of accounting for the specific soundscape when modeling detection performance, suggesting that empirically derived signal-specific detection thresholds must be further adapted to account for differences between real-world ambient environments. Beaches and other coastal environments are an interesting case study for evaluating real-world performance of detection and identification models. In addition to a presenting an unusual and highly dynamic propagation environment for helicopter noise, the natural soundscape resulting from wind and breaking waves has unique spectral and temporal properties that influence how it masks the acoustic signature of helicopters. Here, measurements of a helicopter operating in a beach environment are compared with theoretical and empirical models including models of airborne noise from breaking waves [Bolin and Åbom, J. Acoust. Soc. Am. **127**, 2771–2779 (2010)]. Detection and identification performance are compared with predictions of models for rotorcraft noise—developed in other ambient environments—and models for wind-turbine noise—developed for similar coastal environments.

9:00

5aNS2. Aircraft dose-response research in National Park backcountry areas. Amanda Rapoza, Kristin Lewis, Hastings Aaron, and Cynthia Lee (Volpe National Transportation Systems Ctr., U.S. Dept. of Transportation, 55 Broadway, Cambridge, MA 02142, aaron.hastings@dot.gov)

The Federal Aviation Administration and National Park Service conducted joint research to better understand visitor response to noise from commercial air tour operations over units of the National Park System. Dose-response research in National Parks was conducted during the 1990's at heavily visited, frontcountry sites. To expand this work, aircraft dose-response measurements were recently performed at seven backcountry sites in four National Parks. These sites provided both day and overnight hiking and camping opportunities, with visits ranging from one hour to several days. The sampling approach included side-by-side use of three survey instruments, enabling the evaluation of multiple response variables (annoyance, interference with natural quiet, and acceptability) and comparison of audio clip and *in situ* dose-response, along with the effects of wording and question order. The evaluation of visits longer than previously studied necessitated methodological enhancements including the use of global positioning system based tracking of visitor locations. In total, over 4600 visitor surveys and 50 days of simultaneous acoustic data were collected. This paper, the first of two on this research, describes the research methods and visitor surveys and presents summary results of collected data.

9:15

5aNS3. Aircraft dose-response relations for day-use visitors to backcountry areas in National Parks. Amanda Rapoza (Volpe National Transportation Systems Ctr., U.S. Dept. of Transportation, 55 Broadway, RVT-41, Cambridge, MA 02142, amanda.rapoza@dot.gov), Erika Sudderth (Volpe National Transportation Systems Ctr., Comput. Sci. Corp., Cambridge, MA), Kristin Lewis, Cynthia Lee, and Aaron Hastings (Volpe National Transportation Systems Ctr., U.S. Dept. of Transportation, Cambridge, MA)

The Federal Aviation Administration and National Park Service conducted joint research to better understand visitor response to noise from commercial air tour operations over units of the National Park System. Dose-response relationships developed for heavily visited, frontcountry sites in National Parks showed significant differences in responses between visitors to overlook and short-hike sites, suggesting that activity and visit duration are key influences. An extensive recently collected dataset from backcountry day-use visitors was utilized to further explore these and other influences on dose-response relationships. In this second of two papers on this research, we

describe the model-fitting approach used to identify the combination of noise exposure metrics (dose variables) and mediator variables that best predict visitor responses to aircraft noise. The interpretation and application of the best fit model is presented along with the previously developed front-country model. Such dose-response relationships can be used as a tool for evaluating potential impacts of air tours on visitors to National Parks.

9:30

5aNS4. Study on the aerodynamic noise of centrifugal compressors with rotor static eccentricity. Kaijun Wei, Shuguang Zuo, Huijuan He, and Zhe Wang (Clean Energy Automotive Eng. Ctr., Tongji Univ., No. 4800, Cao'an Rd., Jiading District, Shanghai 201804, China, 1110678@tongji.edu.cn)

Centrifugal compressors are widely used in industrial applications. Rotor static eccentricity is among the common faults in centrifugal compressors, and it has a significant influence on the aerodynamic noise. This paper compares the amplitudes and frequency spectrums of the aerodynamic noise of a centrifugal compressor with no eccentricity and the same compressors with different static eccentricities. The aerodynamic noise is calculated by a hybrid method. Large eddy simulation is applied to solve the unsteady three-dimensional internal flow of the compressor, and the pressure fluctuations obtained by the simulation are considered as the noise sources. The far-field aerodynamic noise generated by the noise sources is predicted using the aeroacoustic finite element method. The results show that centrifugal compressors with eccentricity have higher noise level and different spectrum characteristics.

9:45

5aNS5. Interaction noise reduction for a counter-rotating fan by slitted trailing-edge of the forward rotor. Chen Wang and Lixi Huang (Lab of AeroDynam.. and Acoust., HKU Zhejiang Inst. of Res. and Innovation and Dept. of Mech. Eng., The Univ. of Hong Kong, Pokfulam Rd., Hong Kong, Hong Kong, lixi@hku.hk)

A counter-rotating configuration can decrease the size and weight of fans by eliminating stators and increase the efficiency by recovery of the swirl velocity losses. However, that potential is not fully harnessed due to perhaps the issue of noise. This study explores one idea of passive noise reduction for a small axial-flow counter-rotating fan (120 mm in diameter) by the introduction of slitted trailing-edge for the forward rotor. The fan is designed with simple velocity triangle analyses which are checked by 3D steady-flow numerical simulation and experimental measurements of aerodynamic performance. The aerodynamic consequence and the acoustic benefit of such slit geometry are investigated experimentally when the separation distance between the forward and aft rotor is 4 mm. The results show that there is a reduction of total pressure compared with the baseline fan (without slit) at the same rotational speeds, but this is easily compensated for by slightly raising the rotational speeds. A reduction of about 5 dB in overall noise is achieved for the same aerodynamic output in all directions around the fan center. The spectral comparison at the fan inlet indicates that the most prominent interaction noise peaks are suppressed greatly by such trailing edge slits.

10:00–10:15 Break

10:15

5aNS6. Different experimental methods for the measurement of broadband noise sources in ducts. Kunbo Xu and Weiyang Qiao (School of Power and Energy, Northwestern PolyTech. Univ., No.127 Youyi Rd. Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

Aeroengine broadband fan noise is a major contributor to the community noise exposure from aircraft. It is currently believed that the dominant broadband noise mechanisms are due to interaction of turbulent wake from the rotor with the stator, and interaction of the turbulent boundary layers on the rotor blades with the trailing edge. Two different methods are presented that enable the separation of different broadband noise sources in turbomachinery ducts, one with respect to modal decomposition developed by DLR, and the other with focused beamformer technique developed in university of Southampton. Different measurement mechanisms are displayed to explain the merits, faults, and requirements.

10:30

5aNS7. Experimental validation of a new intensity estimation method. Benjamin Christensen, Derek C. Thomas, Kent L. Gee, Tracianne B. Neilsen, and Menley Stewart (Brigham Young Univ., 715E 700N, Provo, UT 84606, ukeben@gmail.com)

A new method of estimating acoustic intensity has recently been developed in efforts to improve acoustic measurements of launch vehicles [Thomas *et al.*, "Methods for estimating acoustic intensity in rocket noise fields," JASA **134**(5) 4058–4058 (2013)]. This new method, known as the phase and amplitude gradient estimation (PAGE) method, improves upon the traditional finite difference p-p method for estimating acoustic intensity. The primary advantage of the PAGE method is that it allows for accurate intensity measurements over a larger frequency band. Under certain conditions, it is possible to unwrap the phase component of the PAGE method, allowing for accurate intensity estimates well above previous limitations. The advantages and limitations of the PAGE method are investigated experimentally by taking measurements of arrays of loudspeakers. Preliminary uncertainty analyses of both the PAGE and the finite difference p-p methods are also presented.

10:45

5aNS8. The flow-induced sound of a wall-mounted finite airfoil. Danielle Moreau and Con J. Doolan (School of Mech. Eng., Univ. of Adelaide, North Terrace, Adelaide, SA 5005, Australia, danielle.moreau@adelaide.edu.au)

In this study, the aeroacoustic behavior of a wall-mounted finite airfoil is experimentally investigated. Compared to a semi-infinite or two-dimensional airfoil where end effects are not considered, a wall-mounted finite airfoil is more realistic, especially for applications such as wind turbine blades attached to a hub, submarine hydrofoils mounted to a hull, or stators connected to a hub or outer wall. Acoustic and aerodynamic measurements have been taken in an anechoic wind tunnel with single microphones, multiple acoustic beamforming arrays, and hot-wire anemometry. These measurements are used to examine changes in flow topology and radiated noise as a function of Reynolds number and airfoil aspect ratio. Additionally, the data gives insight into the influence of flow at the airfoil tip and wall junction on noise production.

11:00

5aNS9. An experimental research on the wake flow structure of serrated cascade. Kunbo Xu and Qiao Weiyang (School of Power and Energy, Northwestern PolyTech. Univ., No.127 Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

This study concerns the mechanisms of the turbulent broadband noise reduction for cascade trailing edge serrations while the inlet velocity changed from 30 m/s to 50 m/s. The turbulence spatio-temporal information are measured with 3D hot-wire. The experiment is carried out in the Northwestern Polytechnical University low speed open jet wind tunnel. It showed the spreading rate of the wake and the decay rate of the wake centerline velocity deficit increased with serrated edge compared to the straight edge, and the three components of velocity changed differently with serrated trailing edge. It is also found that the turbulence peak occurs further from the airfoil surface in the presence of the serrations, and the serrations widened the mix area, which allowed the flow mixed together ahead of the schedule.

11:15

5aNS10. An equivalent source model for the sound intensity in the vicinity of a high-performance military aircraft. Trevor A. Stout, Gee L. Kent, Tracianne B. Neilsen, Benjamin Y. Christensen, Derek C. Thomas (Phys., Brigham Young Univ., 688 North 500 East, Provo, UT 84606, titorep@gmail.com), and Michael M. James (Blue Ridge Res. and Consulting LLC, Asheville, NC)

The sound field of the F-22A Raptor has been measured extensively using arrays of microphones including an intensity probe to the sideline and rear of the aircraft. Recently, an equivalent source model (ESM) has been proposed that incorporates two arrays of monopole sources placed along the jet axis and their image sources to account for the hard ground. The monopole amplitudes in each case are described by a Rayleigh distribution, and one array includes a phase angle between the monopoles to produce correlated noise steered in a specific direction. The model closely replicates the measured SPL at the majority of measurement locations. However, agreement with energy-based quantities such as the vector acoustic intensity may not follow from agreement with the pressure measurements alone. Hence, in this work, the ESM's ability to produce intensity estimates that match the measured F-22A intensity is evaluated. A new method of estimating acoustic intensity, known as the phase and amplitude gradient estimation (PAGE) method, is applied to the measured F-22A data. The PAGE method accurately estimates intensity in a larger bandwidth than the traditional finite difference method, expanding this discussion of the ESM's accuracy. In addition, sensitivity of ESM parameter selection, e.g., correlated source phase angle and peak amplitude location, on changes in the predicted intensity is examined. [Work sponsored by ONR.]

11:30

5aNS11. Military noise limits: Design criteria versus operational effectiveness. Bruce E. Amrein (Human Res. & Eng. Directorate, Army Res. Lab., Perceptual Sci. Branch, 520 Mulberry Point Rd., Aberdeen Proving Ground, MD 21005, bruce.e.amrein.civ@mail.mil) and Tomasz R. Letowski (Human Res. & Eng. Directorate, Army Res. Lab., Aberedeen Proving Ground, MD)

Noise requirements in military environments differ significantly from typical industrial or occupational situations. Both in combat and in training, mission success requires offensive equipment and weapons to be more lethal and survivable than those used by the adversary. Higher muzzle velocities, heavier projectiles, and more powerful engines result in high levels of both steady-state and impulsive noise and an increased risk of hearing loss for the users. Weapons firing can expose the user to more energy in a single event than typically experienced in a working lifetime of occupational exposure. In addition, military operations require effective speech communication while minimizing auditory detection of equipment by the adversary. Producing material suitable for various forms of military operations requires unique design criteria often exceeding civilian national or international standards. To meet these unique and often contradicting requirements, the U.S. military developed a military design standard for noise limits. This standard (MIL-STD-1474) was last revised in 1997. This paper describes the effort of the U.S. Army, Navy, and Air Force to update the standard to permit production and fielding of military systems designed to maximize Warfighter effectiveness, while minimizing hearing damage caused by their use.

11:45

5aNS12. Development of a digital noise exposure system for research on noise induced hearing loss. Jun Qin (Elec. and Comput. Eng., Southern Illinois Univ. Carbondale, 1230 Lincoln Dr., ENGR-E207, Mail Code: 6603, Carbondale, IL 62901, jqin@siu.edu)

Over 30 million Americans suffer from noise induced hearing loss (NIHL). Many research projects demonstrated that different types of noise, even with equal sound energies, could produce different amounts of hearing loss. In this project, a novel digital noise exposure system has been developed for generating various noise signals (e.g., pure-tone, Gaussian, impulsive, and complex noise). The developed system can be used to study NIHL produced by different types of noise in an animal model. The system could produce impulse noise with peak sound pressure level (SPL) up to 160 dB, which effectively mimics the noise generated by a military weapon (e.g., M-16 rifle). In addition, continuous Gaussian noise with peak SPL up to 140 dB can be created, which is well above the 85 dB recommended exposure limit established by the National Institute for Occupational Safety and Health (NIOSH). The preliminary results of in an animal study showed significant permanent threshold shift (PTS) produced by 90 shocks impulse noise with peak SPL = 155 dB generated by the system. In summary, the digital noise exposure system replicates environmental noise allowing researchers to study hearing loss in a controlled situation.

Session 5aPA

Physical Acoustics: General Topics in Physical Acoustics

David A. Brown, Chair
ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723

Contributed Papers

9:00

5aPA1. Trillium: A thermoacoustic-Stirling refrigerator implementation using an inline topology. Matthew E. Poese, Robert M. Keolian, Robert W. Smith, Eric C. Mitchell, Cory M. Roberts, and Steven L. Garrett (Appl. Res. Lab., The Penn State Univ., PO Box 30, State College, PA 16804, keolian@psu.edu)

To meet a small but growing demand for low global-warming potential refrigeration technologies, a thermoacoustic-Stirling refrigerator has been developed that uses helium as the working gas. Using a new multi-stage topology [Backhaus and Keolian, "In-line stirling energy system," US patent 7,908,856 B2 (2011)], Trillium employs a collinear stack of three linear motors driven by voltage signals phased 120 degrees apart from one another. Above each motor is a thermal core that is comprised of a pair of novel aluminum microchannel heat exchangers on either side of a rolled Kapton® regenerator. The heat exchangers have an equivalent pore size of about a thermal boundary layer thicknesses that allows high effectiveness. The linear motors drive an acoustic wave that travels along the length of the machine. Taken in isolation, each thermal core and the motor-driven piston above and below (sealed using novel plastic flexure seals) forms a single alpha-Stirling machine; taken together Trillium is three Stirling machines working together with, in principle, zero vibration due to a stationary center-of-mass as a result of the motor phasing. The talk will explain the design and present performance measurements. [Work generously sponsored the Advanced Research Project Agency—Energy (ARPA-E) under their BEETIT program.]

9:15

5aPA2. Gedeon streaming suppression in a small scale thermoacoustic-Stirling engine-generator. D. Wilcox and P. Spoor (Chart Inc. - QDr., 302 Tenth St., Troy, NY 12180, douglas.wilcox@chartindustries.com)

Thermoacoustic-Stirling engines, or traveling wave engines, have been shown to convert heat to acoustic power very efficiently (over 30% first-law) in the laboratory. The laboratory prototypes are generally heated from the inside by an embedded electric heater, have a long, bulky resonator, and deliver their work to an acoustic load rather than as electricity, leaving significant challenges required for commercialization unaddressed. The authors are part of a team developing a compact acoustic Stirling engine that is externally heated and is coupled to a pair of linear alternators, dubbed the Thermoacoustic-Stirling Engine-Generator (TaSEG). An important part of this work has been developing a commercially viable means of suppressing Gedeon streaming, a steady flow that circulates in an acoustic engine's toroidal geometry. In the laboratory, this streaming is typically suppressed by either a latex barrier or a "jet pump,'' a special flow element with asymmetric flow resistance, adjusted from the outside of the engine via a rod that passes through the pressure vessel. This work describes the design and testing of a simple, compact, and inexpensive element with multiple jet-pump orifices (the "jet plate''), which can replace the laboratory versions in a commercial engine.

9:30

5aPA3. Bayesian-based model selection and physical parameter estimation of the acoustical properties of rigid-frame porous media. Cameron J. Fackler, Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St, Greene Bldg., Troy, NY 12180, facklc@rpi.edu), Kirill V. Horoshenkov (Dept. of Mech. Eng., Univ. of Sheffield, Sheffield, United Kingdom), and Amir Khan (School of Eng. and Informatics, Univ. of Bradford, Bradford, United Kingdom)

This paper studies an application of Bayesian inference to determine the most appropriate model for the acoustical properties of rigid-frame porous media. The Bayesian framework encompasses two levels of inference: model selection and parameter estimation. Both are based on the posterior probability distribution, which is unique to each combination of acoustic model and material sample. The posterior distribution provides statistically informed guidance on selecting an appropriate acoustic model for a material sample as well as determining the physical parameters of the porous microstructure. Using the nested sampling algorithm, the posterior distribution for each material sample and acoustic model is explored and integrated numerically. This process computes the Bayesian evidence for each model, which is used to select a suitable model for the acoustical properties of the material. The process also provides estimates of the physical parameters, which are used with the selected model for accurate prediction of the acoustical properties of the material.

9:45

5aPA4. Modeling of sound propagation in the vicinity of rigid porous ground by boundary element method. Yiming Wang and Kai Ming Li (Mech. Eng., Ray W. Herrick Labs., Purdue Univ., 140 South Martin Jischke Dr., West Lafayette, IN 47907-2031, wang1679@purdue.edu)

A standard boundary integral equation (BIE) technique takes advantage of the well-known Green's function for the sound fields above an impedance ground. It leads to a simplified solution that enables the development of efficient two-dimensional boundary element numerical codes for computing sound fields above a locally reacting ground. The method is particularly useful if there are multiple reflecting surfaces and the presence of a mixed impedance ground in the proximity of sources and receivers. However, the BIE formulation cannot be applied readily for calculating the sound fields above a non-locally reacting ground because the corresponding Green's function is generally not available. The so-called two-domain approach was frequently used to model the sound propagation inside the porous material by assuming it as a dissipative fluid medium. By matching the particle velocities and pressures at the interface, a two-domain BIE formulation can be developed for modeling the sound propagation above a rigid porous ground. Use of the two-domain approach has a significant impact on the required computational resources. The current paper presents an alternative method aiming to reduce the computational time by exploring the use of an accurate Green's function for the development of a BIE formulation above a non-locally reacting ground.

10:00

5aPA5. Convergence behavior and steady state response of a rib-stiffened, layered plate structure subjected to high frequency acoustic loading. Jeffrey Cipolla (Weidlinger Assoc., Inc., 1825 K St. NW, Ste. 350, Washington, DC 20006, cipolla@wai.com), Patrick Murray, and Kirubel Teferra (Weidlinger Assoc., Inc., New York, Uganda)

An existing analytical, frequency domain solution for wave propagation in coated, ribbed, three-dimensional elastic layered plates excited by acoustic plane waves provides fast solutions for high frequency excitations. The solution methodology, which is found to be numerically unstable under certain conditions, contains an Ansatz for a particular wave number expansion in the direction of periodicity. Evidence is presented to show that the numerical instability is due to the specific choice of the wave number basis. In order to provide a remedy while retaining the positive aspects of the solution methodology, we determine the set of admissible propagating (and attenuating) waves via an eigenvalue analysis. Several approaches exist to determine the admissible waves of structures with periodicity. The Wave Guide Finite Element (WFE) method leads to a two parameter, nonlinear eigenvalue problem, which is difficult to solve. The Scale Independent Element (SIE) formulation results in a two-parameter quadratic eigenvalue problem and overcomes the numerical issues of the WFE method. This study examines available methods in determining the admissible waves and compares them with those established using the dispersion relation. The computed admissible waves are then compared with the aforementioned Ansatz.

10:15–10:45 Break

10:45

5aPA6. An iterative approach to measurement of oblique acoustic absorption coefficient in three-dimensions. Hubert S. Hall (Naval Surface Warfare Ctr. Carderock, 620 Michigan Ave. NE, Washington, District of Columbia 20064, 61hall@cardinalmail.cua.edu), Joseph F. Vignola, John Judge (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC), and Diego Turo (Dept. of BioEng., George Mason Univ., Fairfax, VA)

The measurement of oblique acoustic absorption coefficient remains an ill-defined test. Unlike Kundt's tube (ASTM C384 and E1050) and Sabine room measurements (ASTM C423), there is no uniform test method or standard to follow. *In-situ* or specific application based testing is the industry norm for arbitrary angular absorption coefficient measurement. An iterative approach has been developed for a free/free/free/free panel in an anechoic chamber. Using numerical model predictions of the acoustic field, microphone measurement data are updated to account for a multitude of acoustic phenomena including diffraction from the panel. This method provides a quick measurement option that is flexible and transportable for many applications.

11:00

5aPA7. Infrasound wind noise reduction via porous fabric domes. John M. Noble, W. C. Kirkpatrick Alberts (US Army Reserach Lab., 2800 Powder Mill, Adelphi, MD 20783, kirkalberts@verizon.net), Richard Raspet (National Ctr. for Physical Acoust., Univ. of MS, University, MS), Sandra L. Collier, and Mark A. Coleman (US Army Reserach Lab., Adelphi, MD)

Often, porous hose arrays or complex pipe rosettes are used for wind noise reduction in long-term infrasound monitoring situations. Both of these wind noise reduction schemes perform well in reducing the wind-generated noise at an infrasound sensor, but each can significantly filter the information reaching the sensor. Ideally, a wind noise reduction scheme should have a flat frequency response and a minimal attenuation over the desired frequency band. To that end, three fabrics, two porous and one non-porous, stretched in dome configurations have been investigated for wind noise reduction. Two porous fabric domes are found to perform comparably to a porous hose array but without its amplitude- and phase-altering properties.

11:15

5aPA8. Focused ultrasonic pulsed source for non-contact measurements in air. Frederic Cohen Tenoudji, Dominique Busquet, Jean François Mourey (Institut Jean le Rond d Alembert,MPIA, Sorbonne Universités, UPMC Univ. Paris 06, CNRS, UMR 7190, 2, Pl. de la Gare de Ceinture, Saint-Cyr-l 78210, France, fcohentenoudji@yahoo.fr),

Ultrasonic spark sources have qualities propitious to non-destructive testing of materials in air. They are energetic, and they generate a short acoustic pulse which bandwidth extends from a few hundred Hertz to several hundred kilohertz. It is shown here that by focusing the cylindrically symmetric wavefront generated by the spark with an elliptical mirror, it is possible to concentrate the acoustic energy on the focal line image of the spark. This technique provides a powerful virtual source of small dimension, which may be localized on the surface of the material to be inspected. The performance of this source for non-contact inspection in transmission of composite materials is evaluated by measurements on test samples. The lateral spatial resolution and the high frequency limitation of this technique are determined.

11:30

5aPA9. Experimental research on a sound insulation structure based on embedded Helmholtz resonators. Xinwu Zeng, Dongbao Gao, and Changchao Gong (College of OptoElectron. Sci. and Eng., National Univ. of Defense Technol., Yanzheng St., Changsha 410073, China, xinwuzeng@nudt.edu.cn)

The structures containing Helmholtz resonators (HRs) have been demonstrated to be one of the metamaterials with negative parameters. Due to its local resonance, the one- and two-dimensional structures were indicated having local resonant forbidden band, which are possible ways to attenuate low-frequency acoustic waves and oscillations. At the resonant frequency, the incident acoustic energy is almost localized around the resonator, when a local resonant gap (LRG) appears at the end of the structure. Since the existence of LRG is only associated with the properties of single resonator, a wide band gap can be produced using periodically arrayed HRs with gradually changed parameters. In this structure, neighbor forbidden gaps are overlapped. A sound insulation device is designed and implemented in this paper based on embedded Helmholtz resonators (HRs). This structure is feasible to insulate acoustic waves transmitting above it. Numerical and experimental results both show that a low-frequency wideband insulation area can be formed by circularly arrayed HRs. The working band can be adjusted by changing the resonant frequency of the HRs in each layer.

11:45

5aPA10. Energy flux streamlines versus acoustic rays for modeling an acoustic lens: Energy flow inside and in the focal region for a carbon dioxide filled spherical balloon in air. Cleon E. Dean (Phys., Georgia Southern Univ., PO Box 8031, Math/Phys. Bldg., Statesboro, GA 30461-8031, cdean@georgiasouthern.edu) and James P. Braselton (Mathematical Sci., Georgia Southern Univ., Statesboro, GA)

As an extension of recently published experimental work [Dean and Parker, "A ray model of sound focusing with a balloon lens: An experiment for high school students," J. Acoust. Soc. Am. **131**, 2459–2462 (2012)], a comparison of ray acoustics and wave analysis via energy flux streamlines [Chapman, "Using streamlines to visualize acoustic energy flow across boundaries," J. Acoust. Soc. Am. **124**, 48–56 (2008)] as a means of visualizing the sound field for a positive acoustic lens in the form of a carbon dioxide filled spherical balloon in air is made. The sound field is expanded in the usual Legendre polynomials and spherical Bessel functions [Anderson, "Sound scattering from a fluid sphere," J. Acoust. Soc. Am. **22**, 426–431 (1950)], and the energy flux vectors at points throughout the regions of interest are calculated [Adin Mann III, *et al*., "Instantaneous and time-averaged energy transfer in acoustic fields," J. Acoust. Soc. Am. **82**, 17–30 (1987)]. Then, energy flux streamlines are plotted using Mathematica routines for comparison with conventional acoustical rays, both inside and outside the balloon. Particular attention is paid to the focal region.

Session 5aPP

Psychological and Physiological Acoustics: Potpourri (Poster Session)

Samuel R. Mathias, Chair
Ctr. for Computational Neuroscience and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215

All posters will be on display from 8:00 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 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

5aPP1. A "better ear" listening strategy for improving speech-in-noise understanding in bilateral cochlear implant users. Alan Kan and Ruth Y. Litovsky (Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, ahkan@waisman.wisc.edu)

For bilateral cochlear implant (BiCI) users, understanding a target talker in noisy situations is difficult. Current efforts for improving speech-in-noise understanding have focused on improving signal-to-noise ratio (SNR) by multi-microphone techniques and signal processing, with only moderate improvement in performance. BiCI users typically report having a "better ear" for listening and recent data collected in our lab has shown that they have an asymmetry in speech unmasking performance. This work proposes a novel listening strategy for improving speech-in-noise understanding by combining: (1) a priori knowledge of a "better ear" and having a BiCI user selectively attend to a target talker in that ear; with (2) signal processing that delivers the target talker to the "better" ear and the noisy background to the opposite ear. We compared performance on a speech-in-noise test with and without this "better ear" strategy, using a virtual auditory space created from individualized head-related transfer functions. Subjects showed an improvement of at least 6 dB SNR in the speech reception threshold when using the "better ear" strategy, demonstrating that the strategy can boost speech-in-noise understanding for BiCI users. The novelty of this strategy is that it can be easily applied to other devices aimed at improving hearing.

5aPP2. Preliminary evaluation of a physiologically inspired signal processing strategy for cochlear implants. Jayaganesh Swaminathan (Sensimetrics Corp., 635 Commonwealth Ave., Rm. 320, Boston, MA 02215, jswamy@bu.edu), Raymond L. Goldsworthy, Patrick M. Zurek (Sensimetrics Corp., Malden, MA), Agnès C. Léger, and Louis D. Braida (Res. Lab. of Electronics, MIT, Cambridge, MA)

An approach to developing a cochlear-implant (CI) signal processing strategy inspired by healthy auditory physiology is described, and preliminary perceptual measurements with CI patients are reported. Envelopes were derived from a phenomenological auditory-nerve model (Zilany and Carney, 2010) that has been rigorously tested against physiological responses to both simple and complex stimuli. The model captures several physiological properties associated with nonlinear cochlear tuning including compression, suppression, and neural adaptation that are known to be critical for the representation of speech in quiet and noise. The envelopes were imposed on constant-rate current pulses via a programmable interface. Performance with this "neural" strategy was compared against the listeners' personal processor and a standard encoder (ACE) processor implementation. Sentence, consonant and vowel identification were obtained from 5 CI users in quiet and + 6 dB of continuous speech-shaped noise. Overall, performance with the neural strategy was intermediate between that with subjects' personal processors and the standard ACE processor. The closeness of results with the neural strategy to the listeners' personal processor was viewed as promising as no individual tailoring of parameters were performed, and the subjects had no training with the neural strategy. Results from modeling efforts and future research directions will be discussed. [Work supported by NIH-NIDCD (R43-DC013006).]

5aPP3. Speech understanding in adults and children for sentences with cochlear-implant simulated spectral degradation and shallow insertion depth. Sara Dougherty, Arifi Waked, and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland, 0119E Lefrak Hall, College Park, College Park, MD 20742, saraclaire216@gmail.com)

While previous studies have compared cochlear-implant simulated (i.e., vocoded) speech understanding between adults and children, they have been acute experiments that have not considered long-term adaptation or training effects. Normal-hearing adult listeners demonstrate significant improvement in vocoded speech understanding with training, particularly if there is simulated frequency-to-place mismatch. The purpose of this study was to compare how adults and children adapt to eight-channel sine-vocoded speech with 0, 3, and 6 mm of frequency-to-place mismatch. Twenty adults (>18 yrs) and ten children (8–10 yrs) were trained on vocoded speech understanding over a four-hour period. The stimuli were a closed set of 40 simple words from which five-word nonsense sentences were constructed. Speech understanding was measured in 45-trial blocks where no feedback was provided, followed by 30-trial blocks where visual and auditory feedback was provided. High variability existed within both groups. On average, children performed worse than adults. Over the first five testing/training blocks, children improved at a slower rate than adults. Some children showed minimal improvement over the testing, whereas most of the adults showed noticeable improvement. These results suggest that adults have developed neural mechanisms that can more effectively adapt to and process degraded and frequency-shifted speech.

5aPP4. Discrimination of inconsistent interaural time differences across frequency in simulated bilateral cochlear-implant users. Francisco A. Rodríguez and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland, 0119E Lefrak Hall, College Park, MD 20742, frodri@gmail.com)

When using highly controlled low-rate stimulation, interaural-time-difference (ITD) just-noticeable differences (JNDs) vary greatly across frequency and individual bilateral cochlear-implant (CI) users. It is unclear how across-frequency variability would affect JNDs in multi-electrode stimulation where there are channel interactions from current spread. Therefore, we simulated variable across-frequency ITD sensitivity and channel interactions in 10 normal-hearing listeners presented with a CI simulation that consisted of 100-pulse/s, 1.5-mm bandlimited pulse trains (PTs) in three frequency regions (middle frequency = 7.6 kHz). The simulated electrode spacing was varied by placing the low- and high-frequency PTs $\Delta = 0.75$, 1.5, 3, or 4.5 mm from the middle frequency. Inconsistent ITD sensitivity across frequency was simulated by applying non-zero ITDs in one, two, or three PTs, while the remaining PTs had zero ITD. The pulses in the PTs were simultaneous, nearly simultaneous, or maximally separated. In a left-right discrimination task, JNDs increased for a decreasing number of dichotic PTs. JNDs increased for $\Delta < 3$ mm, suggesting an important role of channel interactions. JNDs increased for non-simultaneous stimulation likely due to a rate limitation effect. The relatively poorer listener performance from the factors in our simulations suggest a means to predict multi-electrode ITD performance in CI users.

5aPP5. Auditory evoked responses to perceived quality of vocoded speech. Chin-Tuan Tan, Elizabeth Glassman, Samuel Oh (Dept. of Otolaryngol., New York Univ., School of Medicine, 550 First Ave. NBV 5E5, NY 10016, Chin-Tuan.Tan@nyumc.org), WenJie Wang, Brett Martin (Speech and Hearing Program, City Univ. of New York, Graduate Ctr., New York, NY), Arlene Neuman, and Mario Svirsky (Dept. of Otolaryngol., New York Univ., School of Medicine, New York, NY)

One of the common uses of vocoded speech is to simulate what speech would sound like when it is processed via a cochlear implant. However, listeners with normal hearing may perceive vocoded speech differently depending on the carrier signals (noise or tone) or channel bandwidths. In this study, we attempted to determine how physiological measures correlate with the quality ratings of vocoded speech that varied in carrier signal and channel bandwidth. Eleven NH subjects rated the perceived sound quality of a spoken sentence processed with noise and tone vocoders (22-channel analysis/synthesis filterbank) whose channels were broadened and narrowed by different Q factors. Each subject was instructed to rate sound quality on a 10 point scale where "10" indicates "clean" and "1" represents "very-distorted". Quality ratings for six different combinations of carrier and Q factor were very consistent across listeners. Auditory evoked potentials (MMN) were obtained from three NH listeners. Vowel stimuli processed through the vocoders with variations in the Q value were used as stimuli. Changes in N1 latency were observed to correlate with the perceived quality ratings of the vocoded speech. [This study was supported by NIH-K25-DC010834 (PI: Tan), PSC-CUNY (PI: Martin), NIH-R01-DC011329 (PI: Neuman and Svirsky), and NIH-R01-DC003937 (PI: Svirsky).]

5aPP6. The precedence effect: Exploring the build-up of echo thresholds in cochlear-implant listeners. Tanvi D. Thakkar (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, 1500 Highland Ave., Rm. 563, Madison, WI 53705, tthakkar@wisc.edu), Andrew D. Brown (Dept. of Physiol. & Biophys., Univ. of Colorado SOM, Aurora, CO), Heath G. Jones, Alan Kan, and Ruth Y. Litovsky (Binaural Hearing & Speech Lab, Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI)

In reverberant environments, listeners rely on early arriving spatial cues to accurately localize sound sources, a phenomenon known as the precedence effect (PE). In deaf individuals fitted with bilateral cochlear implants (BiCIs), this effect is diminished, ostensibly due to the fact that clinical processers do not preserve binaural cues. We have recently demonstrated that BiCI listeners do exhibit aspects of the PE similar to normal-hearing (NH) listeners when binaural stimulation is restored using synchronized research devices. Here, we consider whether BiCI users also demonstrate an aspect of the PE known as "buildup"—enhancement of the PE after a repeated stimulus. BiCI users with demonstrated sensitivity to interaural time differences (ITDs) were tested using dichotic electrical pulses ($\pm$500μs ITD in opposing "lead" and "lag" pulse pairs, with lead-lag delays of 1–64 ms). On each trial, listeners indicated (1) whether one or two locations were perceived (to assess "fusion") and (2) the location perceived (or, given two locations, the "left-most" location perceived, to assess "localization dominance"). Preliminary results indicate that 'buildup' may be atypical in BiCI users, who have experienced years of acoustic deprivation. Lack of adaptation to redundant stimuli may temper the extent of benefit from restored binaural inputs in reverberant environments.

5aPP7. Recovery from forward masking of vowels and consonants: Effects of age and hearing loss. William J. Bologna (Dept. of Hearing and Speech Sci., Univ. of Maryland, 135 Rutledge Ave, MSC 550, Charleston, SC 29425, bologna@musc.edu), Daniel Fogerty (Dept. of Commun. Sci. and Disord., Univ. of South Carolina, Columbia, SC), Jayne B. Ahlstrom, and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Vowels contain slow periodic fluctuations that contrast with aperiodic noise, which may favor perception of vowels over consonants with simultaneous and forward noise maskers. Older adults with or without hearing loss may be poorer at coding these temporal periodicity cues, which may limit their performance. In this study, younger adults with normal hearing and older adults with normal and impaired hearing identified consonants or vowels in the initial or final position of noise-masked syllables. Syllable and masker duration and relative timing ensured that the final 10, 40, or 100 ms of the syllable occurred after masker offset. Individualized spectral shaping minimized confounding effects of reduced audibility in regions of hearing loss. Preliminary results for initial-position phonemes indicate vowels are less susceptible to simultaneous masking than consonants. Recognition of final-position vowels is facilitated by 40-ms delay, whereas final consonants require 100-ms delay or longer for similar improvement. Younger adults benefit most from these delays, older adults with normal hearing benefit less, and older adults with hearing loss benefit least. These findings suggest that age and hearing loss contribute to prolonged recovery from forward masking, and that vowels have greater resistance to forward masking than consonants. [Work supported by NIH/NIDCD and ASHA.]

5aPP8. Modeling individual differences in overshoot: Effects of age, hearing loss, and efferent feedback. Skyler G. Jennings (Commun. Sci. and Disord., Univ. of Utah, 390 S. 1530 E., Salt Lake City, UT 84112, skyler.jennings@hsc.utah.edu), Jayne B. Ahlstrom, and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

The detection of a short sinusoidal probe in simultaneous masking improves as the probe's onset is delayed from the masker's onset. This "overshoot" may be mediated by the medial olivocochlear (MOC) reflex, whose pathway includes spiral ganglion neurons (SGNs), olivocochlear neurons and the outer hair cells (OHCs). Overshoot was measured in younger adults with normal hearing, older adults with normal hearing, and older adults with hearing loss to test the hypothesis that overshoot decreases as components in the MOC reflex pathway are compromised. Overshoot was significantly reduced in older adults, but only those with hearing loss, which is consistent with overshoot depending primarily on the status of the OHCs and only minimally influenced by age-related reductions in SGNs. Thresholds measured when the probe was near the masker's onset showed large differences across listeners, resulting in appreciable individual differences in overshoot. Simulations were generated from a computational model of the auditory system to quantify the contributions of cochlear hearing loss, MOC reflex strength, and detection efficiency to individual differences in overshoot. Preliminary results suggest that cochlear hearing loss and detection efficiency explain the largest portion of the variance in overshoot among adults with normal and impaired hearing. [Work supported by NIH/NIDCD.]

5aPP9. Modulation masking attributes of narrowband and low-noise noise forward maskers in normal-hearing and hearing-impaired listeners. Adam Svec, Peggy B. Nelson (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota,Minneapolis, MN 55455, svecx002@umn.edu), and Judy Dubno (Dept. of Otolaryngol., Medical Univ. of South Carolina, Charleston, SC)

Savel and Bacon (2003) measured detection thresholds for a 4000 Hz pure-tone signal in the presence of a narrowband noise (NBN) or a low-noise noise (LNN) simultaneous masker. The authors asserted that fluctuations in the envelope of the NBN were likely responsible for its increased masking effectiveness. Because modulation detection interference (MDI) is larger for hearing-impaired (HI) than normal-hearing (NH) listeners using modulated simultaneous maskers (e.g., Lorenzi *et al*., 1997) and forward maskers (e.g., Koopman *et al*., 2008), we measured detection thresholds for NH and HI listeners for pure tones in the presence of either NBN or LNN forward maskers that were either 1-ERB wide or 1/3-ERB wide. Based on previous MDI findings (Koopman *et al*., 2008), we predicted larger differences in masked thresholds for the NBN and LNN conditions for HI than NH listeners. These results for detection thresholds for pure-tone signals have implications for interpreting differences in modulated forward masking for NH and HI listeners. [Work supported by grants from NIH/NIDCD.]

5aPP10. Neurometric amplitude-modulation detection threshold measured in the chinchilla ventral cochlear nucleus following sensorineural hearing loss. Mark Sayles, Ann E. Hickox (Dept. of Speech, Lang. and Hearing Sci., Purdue Univ., Heavilon Hall, West Lafayette, IN 47907, sayles.m@gmail.com), and Michael G. Heinz (Dept. of Biomedical Eng., Purdue Univ., West Lafayette, IN)

Amplitude modulation is a common feature of natural sounds and an important cue in audition. Modulation supports perceptual segregation of "objects" in complex acoustic scenes, and provides information for speech understanding and pitch perception. Previous work in our laboratory showed increased modulation gain without change in temporal modulation transfer function (tMTF) bandwidth in auditory-nerve fiber responses to sinusoidal amplitude-modulated (SAM) tones measured in chinchillas with noise-induced hearing loss (HL), compared to normal-hearing (NH) controls. The ventral cochlear nucleus (VCN) provides significant input-output transformations with respect to amplitude-modulation representation, with enhanced spike synchrony to the amplitude envelope in several distinct cell types. We recorded spike times in response to SAM tones with modulation depths between 3% and 100% from all major VCN unit types in anesthetized NH and HL chinchillas. HL animals were previously exposed to 116 dBSPL 500 Hz-centered octave-band Gaussian noise. Spike times were analyzed in terms of synchrony to the amplitude envelope, tMTFs were calculated, and a signal-detection theoretic analysis was used to compute modulation-detection and discrimination thresholds. Results will be related to human perceptual studies, which have shown better modulation-detection thresholds in HL. [Work supported by an Action on Hearing Loss Fulbright Commission scholarship (M.S.), and NIH grant R01-DC009838.]

5aPP11. An algorithm to improve speech recognition in noise for hearing-impaired listeners: Consonant identification and articulatory feature transmission. Eric W. Healy, Sarah E. Yoho (Speech & Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43017, healy.66@osu.edu), Yuxuan Wang (Comput. Sci. & Eng., The Ohio State Univ., Columbus, OH), Frederic Apoux, Carla L. Youngdahl (Speech & Hearing Sci., The Ohio State Univ., Columbus, OH), and DeLiang Wang (Comput. Sci. & Eng., The Ohio State Univ., Columbus, OH)

Previous work has shown that a supervised-learning algorithm estimating the ideal binary mask (IBM) can improve sentence intelligibility in noise for hearing-impaired (HI) listeners from scores below 30% to above 80% [Healy *et al*., J. Acoust. Soc. Am. **134** (2013)]. The algorithm generates a binary mask by using a deep neural network to classify speech-dominant and noise-dominant time-frequency units. In the current study, these results are extended to consonant recognition, in order to examine the specific speech cues responsible for the observed performance improvements. Consonant recognition in speech-shaped noise or babble was examined in normal-hearing and HI listeners in three conditions: unprocessed, noise removed via the IBM, and noise removed via the classification-based algorithm. The IBM demonstrated substantial performance improvements, averaging up to 45% points. The algorithm also produced sizeable gains, averaging up to 34% points. An information-transmission analysis of cues associated with manner of articulation, place of articulation, and voicing indicated general similarity in the cues transmitted by the IBM versus the algorithm. However, important differences were observed, which may guide the future refinement of the algorithm. [Work supported by NIH.]

5aPP12. Optimizing masker phase effects for use in a portable hearing screening tool. Evelyn M. Hoglund, Lawrence L. Feth, Yonghee Oh, and Niall Klyn (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, hoglund.1@osu.edu)

The current study is a continuation of work toward development of a preclinical indicator for noise induced hearing loss. Masked threshold differences produced by Schroeder-phase maskers have been demonstrated with long duration tones at different frequencies (1, 2, and 4 kHz) as well as with IEEE sentences (Summers and Leek, 1998) for normal hearing listeners, but these differences are not found for hearing impaired listeners. Similar results are also found for repeated short duration tone bursts and greater differences occur with systematically changed tone burst frequencies (Hoglund *et al*., 2012). Using the enhanced channel model (Oh, 2013), threshold differences are predicted, and the Schroeder-phase masker characteristics are adjusted to maximize the difference. This leads to a greater range of masker phase effect differences between the positive and negative Schroeder-phase maskers, and should allow greater sensitivity to preclinical hearing threshold changes. These optimized maskers were also applied to a digit triplets test designed for telephone hearing screening (Watson *et al*., 2012). Testing with both the digit triplets and the short duration tone bursts are compared for ease of use and portability, as well as sensitivity to small changes in post-exposure thresholds. [Work supported by the Office of Naval Research.]

5aPP13. Acceptable noise level in bilinguals: Consideration of signal and masker effects. Gabrielly Azcona (Commun. Sci. and Disord., Long Island Univ. Brooklyn Campus, 617 W. 190th St. #2D, New York, NY 10040, gazcona326@gmail.com), Lupe Buten (Commun. Sci. and Disord., Long Island Univ. Brooklyn Campus, Yonkers, NY), and Lu-Feng Shi (Commun. Sci. and Disord., Long Island Univ. Brooklyn Campus, Brooklyn, NY)

This study explores how bilingual listeners' acceptable noise level (ANL) may be affected by the language of the signal, language of the masker, and talkers in the masker. ANL measures how much a listener tolerates background noise while listening to running speech. It differs from conventional speech recognition tasks in that it does not concern one's ability to comprehend speech. Hence, ANLs are expected to be relatively free of bilingual background. In this study, the signal (running passages from New York State Department of Motor Vehicles driving manual) was presented in two languages (English versus Spanish). The maskers (Auditec babbles) were manipulated in language (English versus Spanish) and number of talkers (four versus twelve). Additionally, three groups of 12 listeners participated in the study—monolingual English, Spanish-English bilingual, and Russian-English bilingual. A $3\times2\times2\times2$ mixed, repeated design was carried out with listener group as the between-subjects factor, and signal and masker language and number factors as the within-subjects factors. Preliminary findings reveal a non-significant effect for all four factors; however, a marginally significant four-way interaction (p = 0.048) invites group wise analysis. Results may help establish new clinical approaches in assessing listeners' speech perception difficulty in noise, regardless of language background.

5aPP14. An analysis of the acoustic characteristics of forest sounds. Doo-Heon Kyon and Myung-Jin Bae (Electronics Eng., Soongsil Univ., Hyeongnam Eng. Hall #1212,Sangdo-dong, Seoul 156-030, North Korea, kdhforce@gmail.com)

The purpose of this study is to define the general acoustic characteristics of forest sounds. Thus, large-scale measurements and analyses were

conducted throughout four seasons for three main mountainous areas in Korea. The results showed there were clear differences in the acoustic characteristics, depending on environments and seasons. As acoustic elements of a forest, there are sounds of water from waterfalls and streams and sounds from birds and insects, and even sounds caused by stepping on snow in winter or fallen leaves in autumn have effect on the seasonal acoustic characteristics. The frequency of forest sounds was about 20 dB(A) smaller than typical sounds occurring in a downtown area, and forest sounds showed flat frequency characteristics in general. Especially, it was investigated that the energy ratio of their ultra-high tone sound domains was only 0.1% of that of a downtown area, while the rate of forest sounds was about 50 times more than that of a downtown area, and there were large differences found in the other frequency bands. Based on the results above, such acoustic differences between forests and downtown areas might have effect on the human health in a long-term view.

5aPP15. Psychometric functions of sentence recognition in amplitude-modulated noises. Yi Shen, Nicole Manzano, and Virginia M. Richards (Dept. of Cognit. Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697, shen.yi@uci.edu)

Although it is known that listeners can take advantage of the fluctuations in the temporal envelopes of background noises to improve the recognition of target messages, the amount of benefit is difficult to compare across listeners and experimental conditions. One solution to this difficulty is to estimate the psychometric functions for speech recognition instead of the speech reception threshold alone (i.e., the 50% correct point on the psychometric function). The current study utilized a rapid psychophysical procedure that enabled the robust estimation of the psychometric function for sentence recognition in noise using as few as 20 sentences. Using this procedure, sentence recognition was measured by presenting target sentences in amplitude-modulated noise maskers. In separate conditions, the target intensity (40 or 70 dB SPL) and the masker modulation rate (1–64 Hz) were systematically varied. Manipulating these two stimulus parameters influenced both the speech reception threshold and the slope of the psychometric function. Data collected from ten young, normal hearing listeners indicated that the fluctuating-masker benefit was much more evident at the higher target level and it exhibited non-monotonic dependencies on masker modulation rate.

5aPP16. Bayesian estimation of high-parameter models of the auditory filter. Yi Shen, Rajeswari Sivakumar, and Virginia M. Richards (Dept. of Cognit. Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697, shen.yi@uci.edu)

The Bayesian adaptive procedure proposed by Shen and Richards [J. Acoust. Soc. Am. **134**(2), 1134–1145 (2013)] was extended to allow the rapid estimation of auditory filters that were asymmetric about their peak frequencies. The estimation of the auditory-filter shape (five free parameters) was achieved using single Bayesian adaptive tracks of 150–200 trials (approximately 15 min to complete). During the experimental track, listeners detected a tonal signal presented with either simultaneous or forward maskers. Both types of maskers consisted of two bands of noises, one on each side of the signal frequency. The Bayesian adaptive procedure iteratively updated the parameter estimates following each experimental trial and determined the stimulus that would maximize the gain of information on the following trial. The stimuli were adaptively manipulated along three dimensions: the masker level and the spectral location of the upper and lower masker bands. The proposed procedure allowed the reliable estimation of the auditory-filter shape for naïve normal-hearing listeners. The model predictions replicated the known effect of the masker-signal simultaneity on the auditory filter.

5aPP17. Phase effects using chirp maskers. Niall A. Klyn, Yong Hee Oh, Evelyn Hoglund, and Lawrence L. Feth (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, klyn.1@osu.edu)

Schroeder-phase maskers appear to interact with the phase curvature of the basilar membrane to produce significantly different amounts of masking depending on the direction of the instantaneous frequency sweep (Oxenham and Dau, 2001). Schroeder-phase maskers may not optimally compensate for the phase-curvature of the basilar membrane, which is not assumed to be linear over its length. This study presents a comparison of masking produced by harmonic complexes that differ only in the phase relationships of their components. Four "chirp" complexes were constructed according to Elberling *et al.* (2007), with both positive and negative signs, and were compared to Schroeder-phase signals (Smith *et al.*, 1986). Masking observed with pure-tones at three signal frequencies (1, 2, and 4 kHz) and with spectro-temporal burst signals (Hoglund *et al.*, 2013) is reported at two presentation levels (50 and 60 dB SPL). The results are discussed in the context of Schroeder-phase masking and current understanding of the phase-curvature of the human basilar membrane.

5aPP18. Reliable discrimination thresholds in 17 trials. Michael Stahl (Bioengineering, Northeastern Univ., 226 FR, Boston, MA 02115, stahl.m@meei.harvard.edu), Michael Epstein, and Mary Florentine (Speech-Lang. Pathol. and Audiol., Elec. and Comput. Eng., BioEng., Northeastern Univ., Boston, MA)

This study explores potential weaknesses of the Zippy Estimation by Sequential Testing (ZEST) psychophysical procedure as a means of acquiring rapid psychoacoustic measurements. Although ZEST is promising for this purpose, previous work used *a priori* knowledge to make optimal choices for initial assumptions, a best case scenario for ZEST's performance. Before the ZEST procedure can be employed as a clinical tool, an understanding of how it performs in the absence of a priori information is needed. Specifically, an investigator must choose: (1) the model psychometric function, (2) the starting level, and (3) the number of trials. The present study explores sensitivity to these choices when ZEST is employed in 2AFC, frequency-discrimination tasks. Data for six normal listeners were obtained for a wide range of initial conditions and compared with simulations. These data indicate that even when an inappropriate psychometric function is used, reliable thresholds can be obtained with only 17 trials when the starting level is within a factor of four times the listener's "true" threshold. These results suggest that ZEST combined with a 2AFC paradigm is a promising candidate for rapid and reliable assessment of listeners' discrimination thresholds. [Work supported by NIH/NIDCD 1R03DC009071.]

5aPP19. Effects of level roving and overall level on correlation change discrimination in naïve and trained listeners. Matthew J. Goupell and Mary E. Barrett (Hearing and Speech Sci., Univ. of Maryland, 0119E Lefrak Hall, College Park, MD 20742, goupell@umd.edu)

Sensitivity to interaural differences is remarkably variable across individuals, even highly trained listeners. In experiment 1, we measured sensitivity to changes in interaural correlation in 28 naïve listeners to determine the expected inter-individual variability for this task. Stimuli were 65-dB-A, 10-Hz narrowband noises with a 500- or 4000-Hz center frequency. Stimuli were tested either without level roving, or with ± 5 dB of level roving in an effort to force listeners to attend to binaural width/movement cues rather than binaural loudness cues. At 500 Hz without level roving, the performance for the naïve listeners was much worse than the performance for the highly trained listeners previously reported in the literature. The addition of level roving significantly degraded performance, suggesting a close relationship between detection of decorrelation and loudness. At 4 kHz with and without level roving, none of the naïve listeners could perform the task. In experiment 2, we measured correlation change sensitivity as a function of overall level in trained listeners. Performance improved as overall level increased to about 65 dB-A, then worsened for more intense levels. The data will be discussed in terms of confusions between width and loudness cues, and in terms of Weber's law.

5aPP20. Gradual decay of auditory short-term memory. Samuel R. Mathias (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, smathias@bu.edu), Christophe Micheyl (Starkey Hearing Res. Ctr., Berkeley, CA), and Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

Recent work suggests that once a representation of a visual object is committed to short-term memory, its precision remains fixed until it finally disappears without trace—a phenomenon called "sudden death." In the present study, we investigated whether auditory representations also experience sudden death. In three experiments, listeners discriminated either the pitches or loudnesses of pairs of tones separated in time by up to 10 s. The data were analyzed using a model that allowed us to estimate both the precision of pitch/loudness representations and the probability of sudden death as a function of the interstimulus interval from listeners' psychometric functions. Contrary to recent findings from vision, we found that auditory representations were no more likely to "die" after 10 s than 0.5 s; instead, they declined in precision (or "decayed") gradually over time. The results point to a qualitative difference between how auditory and visual representations are forgotten from short-term memory.

5aPP21. Do normal-reading listeners use "perceptual anchors" in frequency-discrimination tasks? Samuel R. Mathias (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, smathias@bu.edu), Christophe Micheyl (Starkey Hearing Res. Ctr., Berkeley, CA), and Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

A controversial hypothesis about the origins of dyslexia states that dyslexic listeners are poor at frequency discrimination when the standard frequency is fixed from trial to trial because they find it difficult to use "perceptual anchors." This hypothesis tacitly assumes that normal listeners use perceptual anchors during fixed-frequency discrimination, which they cannot do when the standard frequency is random ("roved"). To test this assumption, we measured frequency-discrimination performance in normal listeners for tones separated by silent intervals (ISIs) of up to 10 s. Critically, in either the first or second half of the experiment (counterbalanced across listeners), the frequency of the first tone was the same on each trial ("fixed" conditions); in the other half, the frequency of this tone was random ("roved" conditions). The data were analyzed using a model that estimated the precision of frequency representations. In the roved conditions, frequency representations were less precise at longer ISIs, indicating that they experienced decay. By contrast, frequency representations experienced little decay in the fixed conditions. The results are consistent with the assumption that normal listeners used perceptual anchors in the fixed conditions because, by definition, perceptual anchors should be stored in long-term memory and therefore less susceptible to decay.

5aPP22. Induced loudness reduction as a function of inducer level. Michael Epstein and Mary Florentine (Speech-Lang. Pathol. and Audiol., Elec. and Comput. Eng., Bioengineering, Northeastern Univ., 360 Huntington Ave., 106A FR, Boston, MA 02115, m.epstein@neu.edu)

Induced loudness reduction (ILR) is a phenomenon that occurs when the loudness of a stimulus (test tone) is reduced by the presence of a proceeding sound (inducer tone) that is presented at a level higher than the stimulus. This effect depends on a number of parameters, including inducer level. Whereas a number of studies have examined the effects of inducer level, there is no comprehensive data set examining the effects of a wide range of inducer levels on a wide range of test-tone levels. The present study fills this void by examining the effects of ILR for 500-Hz inducers when matching affected 500-Hz test tones at fixed levels of 5 dB SL and 20, 40, 60, 80, and 90 dB SPL with unaffected 2500-Hz level-adjusted comparison tones. Inducers primarily had an effect on test tones lower than the inducer level. Moderate-level tones exhibited the most ILR. This is consistent with prior studies examining single inducer levels. In addition, low- and moderate-level inducers appear to slightly increase the loudnesses of sounds at higher levels, indicating that the loudnesses of sounds at virtually every sound level within the dynamic range of hearing can be altered by ILR. [Work supported by NIH/NIDCD 1R03DC009071.]

5aPP23. Factors affecting auditory streaming of random tone sequences. An-Chieh Chang, Inseok Heo, Jungmee Lee, Christophe Stoelinga, and Robert Lutfi (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, 1975 Willow Dr., Madison, WI 53706, achang5@wisc.edu)

As the frequency separation of A and B tones in an ABA_{ABA} tone sequence increases the tones are heard to split into separate auditory streams (fission threshold). The phenomenon is identified with our ability to 'hear out' individual sound sources in natural, multisource acoustic environments. One important difference, however, between natural sounds and the tone sequences used in most streaming studies is that natural sounds often vary unpredictably from one moment to the next. In the present study, fission thresholds were measured for ABA_{ABA} tone sequences made more or less predictable by sampling the frequencies, levels or durations of the tones at random from normal distributions having different values of sigma (0–800 cents, 0–8 dB, and 0–40 ms, respectively, for frequency, level, and duration). Frequency variation on average had the greatest effect on threshold, but the function relating threshold to sigma was non-monotonic; first increasing then decreasing for the largest value of sigma. Differences in the sigmas for A and B tones tended to reduce thresholds, but covariance in the A and B tones had little effect. The results suggest that the principles of perceptual organization underlying streaming may differ for predictable and unpredictable tone sequences.

5aPP24. Bidirectional audiovisual interactions: Evidence from a computerized fishing game. Seth Bensussen, Kenny F. Chou (Dept. of Biomedical Eng., Boston Univ., 677 Beacon St., Boston, MA 02215, sethbens@bu.edu), Lenny A. Varghese (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA), Yile Sun (Volen National Ctr. for Complex Systems, Brandeis Univ., Waltham, MA), David C. Somers (Dept. of Psych., Boston Univ., Boston, MA), Robert Sekuler (Volen National Ctr. for Complex Systems, Brandeis Univ., Waltham, MA), and Barbara G. Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

We used a specially designed computer game to examine behavioral consequences of audiovisual integration. Target stimuli (animated fish swimming across the computer screen) were modulated in size and/or emitted an amplitude-modulated sound. Modulations, visual or auditory, were at 6 or 7 Hz (corresponding to "slow" and "fast"). In one game, subjects were instructed to categorize successive fish as "slow" or "fast" based on the auditory modulations; in another game, they categorized fish based on visual modulation rate. In each game, subjects were instructed to ignore input from the task-irrelevant modality. In each game, modulations could be (1) present only in the modality of interest, (2) present and matching in both modalities, or (3) present but mismatched between modalities. While reaction times were similar across games, accuracy was highest when auditory modulation was the basis for categorizing fish. Accuracy and reaction times improved when cross-modal modulation rates matched, and worsened when modulation rates conflicted. Additionally, accuracy was more strongly affected by between-modality congruence/incongruence when subjects attended to visual modulations than when they attended to auditory ones. Results indicate that audiovisual integration is not entirely under volitional control, and that competition between sensory modalities adversely impacts perception in dynamic environments.

5aPP25. Contribution of detailed parts around talker's mouth for audio-visual speech perception. Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Gen Hasegawa (Graduate School of Information Sci., Tohoku Univ., Sendai, Japan), Toru Abe (CyberSci. Ctr., Tohoku Univ., Sendai, Japan), Tomoko Ohtani, Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan), and Tetsuaki Kawase (Graduate School of Biomedical Eng., Tohoku Univ., Sendai, Japan)

In this study, the relationship between speech intelligibility and effects of the parts around talker's mouth was investigated based on the results of audio-visual speech intelligibility tests. As the stimuli, nonsense tri-syllables speech sounds were combined with three kinds of moving image of the talker's face: original face, circumference of the lips (mouth part extracted from

the original face), and masked lips (face without the mouth). The extracted area around the mouth was varied as a parameter in the circumference-of-the-lips conditions. Generated stimuli were presented with speech spectrum noise to the participants. The results showed that intelligibility scores of several phonemes (/h/, /m/, /w/, /b/, /p/) were increased by adding the visual information. Moreover, there was no significant difference between the score of the original-face condition and that of the circumference-of-the-lips condition. Interestingly, significant difference was observed between the score of the audio-only condition and that of the masked-lips-condition. These results imply that not only a mouth but also areas around the mouth without mouth itself provide rich information for speech understanding. Effects of the mouth edges on speech intelligibility are being examined as a next step.

5aPP26. Auditory attention in a dynamic scene: Behavioral and electrophysiological correlates. Hannah R. Goldberg, Inyong Choi, Lenny A. Varghese, Hari Bharadwaj, and Barbara G. Shinn-Cunningham (Auditory Neurosci. Lab, Boston Univ., 677 Beacon St., Boston, MA 02215, hannahrae.goldberg@gmail.com)

The ability to direct and redirect selective auditory attention varies substantially across individuals with normal hearing thresholds, even when sounds are clearly audible. We hypothesized that these differences can come from both differences in the spectrotemporal fidelity of subcortical sound representations and in the efficacy of cortical attentional networks that modulate neural representations of the auditory scene. Here, subjects were presented with an initial stream from straight ahead and a second stream (from either left or right), each comprised of four monotonized consonant-vowel syllables. Listeners were instructed to report the contents of either the first stream (holding attentional focus) or the second stream (switching attentional focus). Critically, the direction of the second stream informed subjects whether to hold or to switch attention. Pilot results suggest that when the lateral angle of the second stream is small, task performance is linked to subcortical encoding fidelity of suprathreshold sound (as measured using brainstem frequency-following responses obtained separately in the same subjects). Using a paradigm that allows simultaneous collection of behavioral measures, FFRs, and cortical responses, here we test whether differences in top-down attentional control explain subject variability when the second stream's lateral angle is large and coding fidelity does not limit performance.

5aPP27. Behavioral and neural measures of auditory selective attention in blast-exposed veterans with traumatic brain injury. Scott Bressler, Inyong Choi, Hari Bharadwaj, Hannah Goldberg, and Barbara Shinn-Cunningham (CompNet, Boston Univ., 677 Beacon St., Rm. 304, Boston, MA 02215, bressler@bu.edu)

It is estimated that 15–20% of veterans returning from combat operations have experienced some sort of traumatic brain injury (TBI), a majority of which are the result of exposure to blast forces from military ordinance. Many of these veterans complain of complications when understanding speech in noisy environments, even when they have normal or near normal audiograms. Here, ten veterans diagnosed with mild TBI performed a selective auditory attention task in which they identified the shape of one of three simultaneously occurring ITD-spatialized melodies. TBI subject performance was significantly worse than that of 17 adult controls. Importantly, the veterans had hearing thresholds within 20 dB HL up to 8 kHz and brainstem responses indistinguishable from those of the controls. Cortical response potentials (measured on the scalp using electroencephalography) from correctly identified trials showed weaker attention-related modulation than in controls. These preliminary results suggest blast exposure damages cortical structures responsible for controlling selective auditory attention (modulating sensory representations in cortex), and represent a step toward developing novel diagnostic methods to assess functional consequences of specific patterns of blast TBI damage in individual patients.

5aPP28. A bilingual advantage in the irrelevant speech effect. Josh Dorsi (Psych., UC Riverside, 34875, Winchester, NY 92596, jdors002@ucr.edu), Dominique Dominique Simmons, Theresa Cook, Lawrence Rosenblum (Psych., UC Riverside, Riverside, CA), and Oksana Laleko (English, SUNY New Paltz, New Paltz, NY)

The bilingual advantage refers to superior performance by multilingual over monolingual individuals on a variety of linguistic, and non-linguistic cognitive tasks. Tasks that include the bilingual advantage are ones that involve attention control, such as the Simon or Stroop tasks, but not tasks that involve short-term memory (Bialyskok, 2009). The Irrelevant Speech Effect (ISE) is the finding that serial recall accuracy of visual list items declines when task irrelevant speech backgrounds are present (relative to the disruption of white noise backgrounds: Colle and Welsh, 1976; Jones and Macken, 1993). The theories that have been offered to account for ISE are of two kinds, those that assume recall disruption is a function of limited memory, and those that assume it is the result of diffused attention (Elliot, 2002). As the bilingual advantage is found in tasks of attention, but not those concerning memory, a bilingual advantage in the ISE would support an attention based theory. In order to test this prediction, bilingual and monolingual subjects performed serial recall tests against speech and white noise backgrounds. Preliminary results suggest that the size of ISE is substantially smaller in bilingual, than it is in monolingual participants supporting attention-based theories of the ISE.

5aPP29. Air conduction and bone conduction in aging mice. David Chhan (Speech and Hearing BioSci. and Technol., Harvard-MIT Health Sci. and Technol., 1200 Massachusetts Ave., Apt 53E, Cambridge, MA 02138, dchhan@mit.edu), Melissa L. McKinnon (Eaton Peabody Lab., Massachusetts Eye and Ear Infirmary, Boston, MA), and John J. Rosowski (Dept. of Otology and Laryngology, Harvard Med. School, Boston, MA)

Umbo velocity and Auditory Brainstem Response (ABR) measurements in aged mice suggest that there are functional changes in both the inner ear and middle ear for frequencies from 5 to 13 kHz (Doan *et al*., 1996). In this work, we use a combination of air conduction (AC) and bone conduction (BC) stimulation to better quantify the middle ear contribution to age-related hearing loss seen in mice. ABRs were recorded with AC and BC stimuli from BALB/c mice of four different age groups (1, 2, 8, and 12 months); mice of this strain are widely used as models for age-related hearing loss. Results show the threshold stimulus levels for both AC and BC increase as the mice get older, consistent with age-related hearing loss. At frequencies below 12 kHz, the age-related changes in thresholds for all age groups are similar for both stimuli: the AC-BC difference (the air-bone gap) is not statistically significant. This suggests in this frequency range, the hearing loss is primarily sensorineural. At 16 kHz, the air-bone gaps of the two oldest groups are statistically significant suggesting the middle ear contributes to the hearing loss. Thresholds at higher frequencies were not measurable in the two oldest groups.

5aPP30. Can frequency discrimination be indexed by electrophysiological measures. Wen-Jie Wang, Brett A. Martin, Glenis R. Long (Ph.D. Program in Speech-Language-Hearing Sci., The Graduate Ctr., City Univ. of New York, 365 Fifth Ave., New York, NY 10016, wwang2@gc.cuny.edu), and Chin-Tuan Tan (Dept. of Otolaryngol., New York Univ. School of Medicine, New York, NY)

Acoustic change complex (ACC) responses to quadratic frequency glides following a steady pure tone were obtained from eight normal-hearing listeners. The base frequencies were 500 and 1000 Hz. The glides either increased or decreased in frequency by 200 Hz with durations of 50, 100, and 200 ms. The N1 response to stimulus onset and glide onset were evaluated to determine if the elicited ACC was associated with the frequency change in the glides. In contrast to the onset N1, the ACC N1 was delayed.

This delay was used to index the time for frequency glides to reach a potentially detectable frequency change. ACC N1 latency was longer for glides of longer duration. However, the instantaneous frequency at ACC N1 peak latency was relatively constant regardless of glide duration. The difference in ACC N1 latency across glide durations was diminished when expressed as stimulus frequency, ratio of frequency change to base frequency, or associated place shift along the basilar membrane. There was also an effect of base frequency. ACC N1 latency was evoked when the percentage change from the base frequency exceeded ~0.2%. There was no clear evidence that the ACC is dependent solely on the duration or rate of frequency change.

5aPP31. A computational study on different flow patterns around the inner hair cell stereocilia. Srdjan Prodanovic, Sheryl M. Gracewski (Dept. of Mech. Eng., Univ. of Rochester, 383 Quinby Rd., Rochester, NY 14623, s.prodanovic@rochester.edu), and Jong-Hoon Nam (Dept. of Mech. Eng., Dept. of Biomedical Eng., Univ. of Rochester, Rochester, NY)

The mechano-transduction of the mammalian cochlea occurs in the micro-fluid domain between the tectorial membrane and the reticular lamina called the subtectorial space. The subtectorial fluid bathes the bundled stereocilia of the inner hair cells (IHCs). These cells are responsible for the onset of neural impulses in the auditory nerve fibers. Despite the generally accepted postulation that the IHC stereocilia are deflected by shear flow between the two layers, there have been suggestions that other flow modes exist besides the shear flow. We developed a computational model of fluid dynamics in the subtectorial space. The model simulates IHC mechano-transduction excited by different flow patterns. In order to compare different modes of fluid dynamical stimulation, the power efficiency of IHC mechano-transduction was introduced (dissipated power normalized by IHC mechano-transduction current). Besides different flow patterns, the effect of mechanical parameters (such as the gap size between the stereociliary tip and the tectorial membrane, stereociliar bundle stiffness) were investigated. The results demonstrate that the power efficiency for the IHC mechano-transduction depends on the flow pattern in the subtectorial space.

5aPP32. Effect of spatial discontinuity on short-term memory for melodic sequences. Audrey S. Wang, Lenny A. Varghese, Samuel R. Mathias, and Barbara G. Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, Audrey_Wang@buacademy.org)

We investigated whether spatially separating segments of a tone sequence split the sequence into separate perceptual streams, and whether it affected how pitch information was stored in memory. Listeners heard a sequence of six 300-ms complex tones presented either to a single ear, or with the first three and last three tones presented to different ears. They reported whether a subsequent monaural probe tone matched a tone in the original sequence, based solely on pitch (disregarding the ear to which the probe was played). Overall, performance was higher when the probe tone matched either the first or one of the last two tones in a target compared to when the probe matched more interior target elements. For single-ear target sequences, presentation of the probe in the opposite ear diminished performance compared to when it was presented in the same ear as the target. Spatially splitting the target sequence had very little effect. The results suggest that the tone sequences were perceived as a single stream, irrespective of whether there was spatial discontinuity. However, a probe in an unexpected location confused listeners, perhaps causing a spatial attention shift that diminished the already-weak memory representation of the interior tones' pitches.

5aPP33. Auditory cortex membrane potential dynamics in mice during challenging and ethologically relevant natural soundscapes. Mathew J. McGinley, Gregg A. Castellucci, and David A. McCormick (Neurobiology, Yale Univ., 333 Cedar St., New Haven, CT 06510, matthew.mcginley@yale.edu)

Natural sound environments present numerous spectro-temporally complex and highly non-linear sound sources that an animal must assess for relevance. The auditory cortex is thought to be important for the execution of challenging sensory processing tasks. Yet, recordings in the auditory cortex have largely been studied in response to simple or synthetic sounds, or single well-isolated natural sound patterns such as con-specific vocalizations. In particular, membrane potential s in auditory cortex have only been recorded in response to simple stimuli, to which they respond with large rapid depolarization from a stable hyperpolarized baseline voltage. To help determine the dynamics of membrane potential fluctuations in natural conditions, we played natural soundscapes to head-fixed awake mice on a cylindrical treadmill while recording membrane potentials of pyramidal neurons the primary auditory cortex of mice. Ambient sounds were recorded from forests or meadows, particularly near creeks and during sunrise or sunset when numerous species were active. Sound frequencies encompassing the entire hearing range of mice (1–250 kHz) were recorded using a AviSoft UltraSoundGate recording system with CM16/CMPA microphone.

5aPP34. Parameter fitting of a lumped parameter middle ear model. Charlsie Lemons and Julien Meaud (Mech. Eng., Georgia Inst. of Technol., 1035 Hampton St, Atlanta, GA 30318, charlsielem@gmail.com)

Measurements of otoacoustic emissions in the ear canal are affected by the forward and reverse pressure transfer functions and the reverse middle ear impedance. Thus, a comprehensive middle ear model should reflect each of these measurements of middle ear function using a single set of parameters. A middle ear circuit model consisting of a lumped parameter representation of the ossicular chain coupled to a transmission line model of the eardrum was selected (O'Connor and Puria, JASA, 2008). In the original model, the parameters were fit solely to measurements of the magnitude and phase of the stapes velocity relative to the ear canal pressure in the forward direction. In this paper, the model parameters are optimized through a non-linear least square regression method so that the model's forward pressure transfer function, reverse pressure transfer function, and reverse middle ear impedance agree with experimental data for the human middle ear from 0.1 to 10 kHz. This parameter fitting procedure is repeated using experimental data for the guinea pig and gerbil middle ear. Differences in model parameters between the three species are discussed.

5aPP35. Aligning digital holography images of tympanic membrane motion. Jeremie Guignard, Jeffrey T. Cheng, Michael E. Ravicz, and John J. Rosowski (Eaton-Peabody Lab., Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA 02114-3096, jeremie_guignard@meei.harvard.edu)

Modern studies of sound-induced tympanic membrane (TM) motion include digital holography and optical-coherence tomography. These techniques generate high-definition maps of the motion of human and animal TMs that define and quantify spatial patterns of motion. In order to compare such patterns quantitatively between multiple specimens, a common coordinate system is necessary. Differences in the relative position of the sensor and the measured sample induce differences in magnification, shape, and location of the resulting TM image. A registration algorithm was implemented as follows: (1) The plot of the phase of motion at low frequencies was filtered with a moving 2D variance filter to differentiate the membrane from the bone in both reference and moving images (a process called segmentation). (2) A routine calculated the covariance matrix of the segmented images to measure its second central moment. (3) The images were aligned and warped on the basis of their centroid and moment. The method is about ten times faster than state-of-the-art intensity based registration, and the average squared errors of the pixel intensities after registration are not statistically different. The method is an efficient strategy to align TM surface images.

5aPP36. Investigating perceptual structure of concert hall ambiances. Aparna Kanakanahalli Nagendra and SungYoung Kim (TeleCommun. Eng. Technol., Rochester Inst. of Technol., 148 Crittenden Way Apt., #6, Rochester, NY 14623, axk9226@rit.edu)

With the advent of ultra high definition (UHD) video, a new sound reproduction system incorporating height channel to enhance audio-visual interaction is getting researchers' attention. To better understand influence of the contents for the height channel(s) on perceived sound quality, the authors have conducted a listening test wherein listeners compared various concert hall ambiances. The authors first recorded the ambiances of a hall at

175 positions (in various heights and positions) using a MIDI-controlled piano. Subsequently we investigated interrelation of ambiances with subordinate objectives: (1) reveal perceptual space of the ambiances and (2) unfold latent yet salient factors. We selected 10 ambiances placed at three-dimensional equidistant from the source as the stimuli. Fifteen musicians participated in a subjective evaluation where they reported perceived dissimilarity between two randomly presented stimuli. The dissimilarity matrices were then analyzed via INDSCAL and the results showed that the stimuli could be coordinated along two bases which corresponded to distance from the source and spectral kurtosis of the stimuli respectively. Finding those salient bases will help assist spatial audio researchers in rendering height ambiances with smaller yet relevant subset of ambiances.

5aPP37. Effects of spectral degradation on attentional modulation of auditory responses to continuous speech. Ying-Yee Kong (Dept. of Speech Lang. Pathol. & Audiol., Northeastern Univ., 226 Forsyth Bldg., 360 Huntington Ave., Boston, MA 02115, yykong@neu.edu), Ala Mullangi (BioEng. Program, Northeastern Univ., Boston, MA), and Nai Ding (Dept. of Psych., New York Univ., New York, Massachusetts)

Auditory attention enhances phase-locked responses to the temporal envelope of the attended speech stream in a competing background. This study investigates how cortical responses are affected by spectral degradation and the extent to which low-frequency fine-structure cues facilitate sound segregation. Two competing speech streams were presented diotically to normal-hearing subjects. Each subject was tested with unprocessed speech, vocoder speech (8 to 64 channels), and six-channel high-frequency vocoder + low-pass filtered speech simulating electric-acoustic stimulation (EAS). Ongoing EEG responses were measured in each condition. Cross-correlation between speech envelope and EEG responses was calculated at different time lags. For unprocessed speech, the cross-correlation function showed opposite signs of correlations between the attended and unattended speech, supporting the neural mechanism of suppression of the competing speech stream. This suppression mechanism, however, was only evident for the 32- and 64-channel vocoder conditions, but not for the 8- and 16-channel conditions. This indicates that greater frequency resolution is required for sound segregation. For EAS, the pattern of cortical responses was similar to that of the eight-channel vocoder condition, although speech intelligibility was higher in EAS than vocoder speech. Consistent with previous psychoacoustic evidence, these neural results further argue against the segregation mechanism for EAS benefit.

FRIDAY MORNING, 9 MAY 2014 — 553 A/B, 9:00 A.M. TO 12:00 NOON

Session 5aSA

Structural Acoustics and Vibration: Recent Advances in Structural Acoustics and Vibrations

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Contributed Papers

9:00

5aSA1. Efficient analysis of dynamic coupling between modifications to complex systems. Andrew Wixom and J. Gregory McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, awixom@bu.edu)

This paper outlines a strategy for identifying and exploiting the lack of dynamic coupling between potential modifications to a vibroacoustic system. Often a designer develops a set of modifications and wishes to determine those that perform the best relative to some metric, in particular, we consider performance metrics that rely on the steady state forced response of the structure over a frequency band of interest. In this case, the effects of coupling are shown to appear as a force residual and therefore may be neglected if sufficiently small. When modifications are uncoupled from one another, the computational cost of analyzing all possible modified systems can be greatly reduced. For example, if up to 4 of 7 potential modifications are allowed to be applied to a system, there are 98 possible modified systems that need to be analyzed to determine the optimal combination. However if all of the modifications are found to be uncoupled, then only seven modified systems need to be analyzed to generate the results for all possible modified systems. An example problem demonstrates the benefits of uncoupling between modifications.

9:15

5aSA2. Resonant vibration control of a cylindrical shell using distributed tuned mass dampers. Christopher Page (Noise Control Eng., Inc., 799 Middlesex Turnpike, Billerica, MA 01821, c.page@noise-control.com), Peter Avitabile, and Christopher Niezrecki (Structural Dynam.. & Acoust. System Lab., Univ. of Massachusetts - Lowell, Lowell, MA)

Often structure borne noise has a deleterious effect on the performance and aesthetics of many commercial and military systems. Solutions to structural noise problems can be broadly classified into three categories: active, semi-active, and passive. With ever increasing improvements in electronics and computing, much of the recent research emphasizes active and semi-active solutions. However, passive solutions remain favorable in practice as they are often less costly to implement and less imposing on the system architecture. The tuned mass damper (TMD) is a common passive approach. Tuned mass dampers are generally used to affect a single undesirable resonance or forcing frequency. In order to affect multiple resonances multiple tuned absorbers are typically required, causing the design and implementation to be prohibitively difficult. This paper develops the concept of a modally enhanced dynamic absorber (MEDA) as a dynamic design philosophy, whereby the mass of peripheral equipment can be utilized and employed in attenuating the structural response of a larger system in some

5a FRI. AM

desirable way. The concept is demonstrated numerically and experimentally on a steel cylindrical shell. The importance of TMD attachment location with respect to the mode being de-tuned, robustness of TMD damping, and the effect of TMD mistuning is studied.

9:30

5aSA3. Transient response of flat-panel distributed mode loudspeakers. David Anderson, Stephen Roessner, and Mark F. Bocko (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, dander22@hse.rochester.edu)

The acoustic impulse response of a loudspeaker must have short rise and decay times to provide accurate reproduction of signals with fast transients, such as human speech. In a flat panel distributed mode loudspeaker (DML), the transverse bending waves that are the primarily source of acoustic radiation, propagate across the panel at finite velocity. Therefore, regions of the panel more distant from the panel driving point(s) radiate sound at later times, which broadens the DML's acoustic impulse response. Furthermore, bending wave propagation in a rigid panel is dispersive, with a wave velocity that is proportional to the square root of the frequency, which creates additional distortions of signals with fast transients. Mechanical simulations and scanning laser vibrometer measurements of glass panel DML's clearly illustrate the delayed radiation and dispersion distortion mechanisms. Furthermore, measurements of the acoustic impulse response of DML's are in agreement with simulations, which provides additional evidence that the identified mechanisms are a major source of audio distortion in DML's.

9:45

5aSA4. Audio frequency response of glass flat panel distributed mode loudspeakers. Stephen Roessner, David Anderson, and Mark F. Bocko (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, stephen.roessner@rochester.edu)

The development of glass flat panel distributed mode loudspeakers (DML's) will enable compelling new applications, such as flat panel displays that double as loudspeakers, or in architectural and automotive applications, window glass that can produce sound or actively cancel environmental noise. In this paper we discuss the frequency response of glass flat-panel distributed mode loudspeakers. The density of panel bending modes, their quality factors (Q's), and the modal radiation efficiencies determine a DML's frequency response. For example, a 0.55 mm thick cover glass for a 55 in. television has more than 10 000 bending modes in the audible frequency range. The density of modes increases with frequency and above a threshold frequency, determined by the panel dimensions and the mode Q's, the mode spacing becomes less than the width of the individual modes, so the overall frequency response approaches the smooth, flat response of an ideal pistonic loudspeaker. However, below this frequency, discrete modes produce prominent peaks in the frequency response. We present simulated and measured glass panel frequency responses, including the effect of a thin layer of trapped air behind a DML panel, and we discuss the need for high internal friction (low Q) glass.

10:00

5aSA5. An analysis of the weighted sum of spatial gradients (WSSG) control metric in active structural acoustic control. Yin Cao, Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N225 ESC, Provo, UT 84602, caoyfive@gmail.com), Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT), and Pegah Aslani (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The choice of the control metric used and how it is measured is an important factor in active structural acoustic control (ASAC) and has been a topic of research for many years. In ASAC, the objective is to minimize the radiated power, but this metric is difficult to implement in practice. Previous research has identified a control metric named the weighted sum of spatial gradients (WSSG) that has demonstrated effectiveness in attenuating the radiated power from a plate. This metric only requires local vibration information on the plate and hence is easier to implement than previous control metrics. It is also largely insensitive to boundary conditions. In addition, ASAC using WSSG has been demonstrated to be generally more effective than using volume velocity as the control metric. This paper demonstrates that WSSG closely approximates the ideal of using the radiated sound power as the control objective. It will be shown that the weights used to implement WSSG are also very important, as they directly impact the control effectiveness. Both theoretical and experimental results show that optimum weights exist to implement WSSG. These results suggest that WSSG has the potential ability to be used in more practical and complex structures.

10:15–10:30 Break

10:30

5aSA6. Dynamic response of cross-linked random fiber networks. Sahab Babaee (SEAS, Harvard Univ., 29 Oxford St., Pierce Hall 410, Cambridge, MA 02138, sbabaee@fas.harvard.edu), Ali Shahsavari, Catalin Picu (Dept. of Mech. Eng., Rensselaer Polytechnic Inst., Troy, NY), and Katia Bertoldi (SEAS, Harvard Univ., Cambridge, MA)

Systems composed of fibers are ubiquitous in the living and artificial systems, including cartilage, tendon, ligaments, paper, protective clothing, and packaging materials. Given the importance of fiber systems, their static behavior has been extensively studied and it has been shown that network deformation is nonaffine for compliant, low-density networks and affine for stiff, high-density networks. However, little is known about the dynamic response of fibrous systems. In this work, we investigate numerically the propagation of small-amplitude elastic waves in these materials and characterize their dynamic response as a function of network parameters.

10:45

5aSA7. Optimization of spherical numerical integration methods in reconstruction of sound field with spherical microphone array. Minzong Li, Jiangming Jin, and Huancai Lu (College of Mech. Eng., Zhejiang Univ. of Techonolgy, 18#, Chaowang Rd., Hangzhou 310014, China, liminzong_svlab@163.com)

Choosing an effective spherical numerical integration method is one of the key problems in reconstruction of incident sound field by using spherical microphone array based on spherical near-field acoustic holography (SNAH). Many different spherical numerical integration methods are reported, such as I.H.Sloan, Spherical-t design, J.Fliege, etc., each method applies different distributions of discrete positions and integral weights in the design of spherical microphone array. In the paper, the accuracies of sound pressure reconstruction results based on SNAH with different spherical numerical integration methods are examined and compared. It is found that spherical-t integration method can provide more options of discrete points, simple algorithmic due to equal weight of each point, and the same or higher precision of sound pressure reconstruction. Meanwhile, the expansion terms of reconstructed sound pressure has to be truncated at an optimal number in order to minimize the impact of spherical numerical integration error, the approach to choosing the optimal truncated term number with respect of the number of microphones in spherical numerical integration is discussed in details.

11:00

5aSA8. Patch nearfield acoustic holography based visualization of spatial distribution of sound quality objective parameters. Jiangming Jin, Fang Yuan, and Huancai Lu (College of Mech. Eng., Zhejiang Univ. of Techonolgy, 18#, Chaowang Rd., Hangzhou 310014, China, jjm@zjut.edu.cn)

Psychoacoustic research reveal that sound pressure level reduction in cabin does not mean improvement of human's subjective auditory perception. This paper presents Patch Nearfield Acoustics Holography-Sound Quality (Patch NAH-SQ) combined analysis method with the advantages of NAH and SQ. This methodology breaks the limit of measurement aperture of conventional NAH and can visualize of spatial distribution of sound pressure and sound quality objective parameters of entire sound field by using a small number of microphones. From the 3D images of spatial distribution of sound pressure and sound quality objective parameters, the sound sources which most contribute to human's auditory perception can be located. In this paper, the impact of the choice of parameters such as overlap ratio of

Patch holographic surface, SNR, stand-off distance and frequency on reconstruction error of sound pressure and sound quality objective parameters is examined. Finally, the experiment of Patch NAH-SQ methodology proposed is validated in anechoic camber, the results are presented and analyzed.

11:15

5aSA9. A fully coupled finite element-boundary element vibro-acoustic analysis for laminated composite structures with enclosed acoustic cavities. Atanu Sahu, Arup Guha Niyogi (Dept. of Civil Eng., Jadavpur Univ., 188 Raja S.C. Mallik Rd., Kolkata, West Bengal 700032, India, sahuatanu@daad-alumni.de), Michael Rose (Inst. of Composite Structures & Adaptive Systems (Adaptronic Devision), German Aerosp. Ctr. (DLR), Braunschweig, Germany), and Partha Bhattacharya (Dept. of Civil Eng., Jadavpur Univ., Kolkata, India)

A major contribution of aircraft cabin noise is due to the vibrating airframe structure in which the use of lightweight composite materials is quite common. Due to the strong influence of such vibrating structures on the enclosing fluid, such as air, and vice-versa, a fully coupled analysis involving both the systems is necessary in many cases. Hence the present work details a fully coupled vibro-acoustic analysis of such type of flexible cavity subjected to the external mechanical excitation based on a coupled finite element (FE) and boundary element (BE) approach. The structural mode shapes and eigen frequencies are computed using the FE free vibration analysis of *in-vacuo* structures, which are subsequently coupled through the mobility equation with the acoustic part modeled with the BE method. The developed approach is also well compared with the fully coupled FE approach but the advantage derived here is that it is sufficient to model each domain independently thus reducing the problem dimension. The discussed method ensures full coupling between the two systems considering the cavity acoustic back pressure and also enables a fully coupled analysis of a double-walled structure attached to an acoustic cavity which resembles a true aircraft cabin.

11:30

5aSA10. Optimal research on volume velocity-matching wave superposition method based on tripole. Shaowei Wu and Yang Xiang (School of Energy and Power Eng., Wuhan Univ. of Technol., Peace Ave., Wuhan, Hubei Province, No. 1040, Wuhan, Hubei 430063, China, thinkwsw@qq.com)

A volume velocity-matching wave superposition method for sound prediction of vibrating structure based on tripole is proposed. In the method, the position of the equivalent sources has similar geometry shape with radiating structure using a scale coefficient K. The calculating cut-off frequency of structure under a meshing pattern is predicted by setting 1.5% volume velocity relative error limit using surface vibration velocity. The cut-off frequency is defined as the maximum value of all wavenumber ka, at which the relative error just exceeds 1.5% when K iterates through (0,1) with step K. Then, for calculating frequency range [k1,k2], the optimal position of the equivalent sources, in which the relative error of volume velocity is minimum, is determined in the calculating cut-off frequency range by searching K in set $\{K|K,ka\geq k2\}$. The transfer matrix between pressure and surface volume velocity is constructed in the optimal position by using tripole as equivalent source. After that, the sound radiation of structure is predicted to get high precision prediction. The results of numerical simulation show that the method is good at predicting the sound radiation of structure and the predicting error is very low within the cut-off frequency range once the optimal position is determined.

11:45

5aSA11. Guided modes with multiple zero-group-velocity points in fluid-filled cylindrical pipes. Hanyin Cui, Weijun Lin, Hailan Zhang, Xiuming Wang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21, Beisihuanxi Rd., Beijing 100190, China, cuihanyin@mail.ioa.ac.cn), and Jon Trevelyan (School of Eng. and Computing Sci., Durham Univ., Durham, United Kingdom)

It is known that guided modes in isotropic hollow cylinders exhibit backward wave propagation with negative group velocity. And interferences between backward and forward waves generate zero-group-velocity (ZGV) resonances with a finite wavenumber but vanishing group velocity. These ZGV resonances can be applied for non-destructive evaluation (NDE) of hollow pipes. In this paper, influences of a fluid-loading on ZGV resonances in pipes are studied for the possible application of integrity inspections of oil transportation pipelines. From numerically simulated frequency-wavenumber spectra of axisymmetric guided modes, in addition to the backward mode with a single ZGV point, certain branches change the sign of their slopes for twice (i.e., two ZGV points in one branch). Such multiple ZGV modes might be caused by the strong repulsion between the backward mode with a single ZGV point that is propagating in the hollow pipe and a number of longitudinal modes in the fluid cylinder. It is found that, from wave structure analyses, ZGV points correspond to relatively large displacement amplitudes at the pipe's inner and outer interfaces. It indicates that guided modes with multiple ZGV points can be sensitive to the surface features of fluid-filled pipes, which is useful for NDE application.

Session 5aSC

Speech Communication: Speech Perception II (Poster Session)

Neal P. Fox, Chair
CLPS Dept., Brown Univ., Box 1821, Providence, RI 02912

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors the opportunity to see other posters, author of odd numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and authors of even numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

5aSC1. Infants' perception of source size in vowel sounds. Matthew Masapollo, Linda Polka (McGill Univ., 1266 Pine Ave. West, Montreal, QC H3G 1A8, Canada, matthew.masapollo@mail.mcgill.ca), Athena Vouloumanos (New York Univ., New York, NY), and Lucie Ménard (Univ. of PQ at Montreal, Montreal, QC, Canada)

Recent research shows that pre-babbling infants can recognize infant-produced vowels as phonetically similar to adult and child vowel productions (Polka *et al*., submitted), indicating that infants normalize for speaker size information. Yet little is known about whether infants encode information about speaker size in speech sounds and then use this information for source identification. Here, we investigate whether infants preferentially attend to smaller visual objects over larger visual objects when they hear infant speech sounds, and attend to larger visual objects over smaller visual objects when they hear adult speech sounds. We are currently testing 9-month-old infants using an intermodal matching procedure, in which they are presented with isolated vowel sounds synthesized to emulate productions by either adult female or infant speakers, along with side-by-side geometric shapes that differ in size (e.g., a large square and a small square). Preliminary analysis suggests that infants display greater mean proportion-looking times to the congruent shape-voice pairs, but additional data collection is ongoing. The implications of these findings for theories of infant speech perception will be discussed.

5aSC2. Interaction of memory and specificity in auditory repetition priming. Georgia Zellou and David Embick (Linguist, Univ. of Pennsylvania, 800 N. 48th St., #26, Philadelphia, PA 19139, gzellou@sas.upenn.edu)

This study examines the influences of abstract and episodic representations of words in auditory repetition priming. Two manipulations of prime and target items were employed, token-change and voice-change. First, either the target item consisted of a different token spoken by the same speaker as the prime (token-change), or the target item was a token spoken by a different speaker/different gender (voice-change) than the prime. Second, lapse between prime and target varied across three conditions: no-lapse (no intervening trials between prime and target), medium-lapse (exactly ten intervening trials), and long-lapse (exactly 20 intervening trials). Stimuli were presented using an auditory lexical decision task (with an equal number of English-like nonwords). The reaction time data reveal the greatest facilitation between token-change no-lapse prime-target pairs (256 ms), followed by voice-change no-lapse pairs (182 ms). Facilitation effects were comparable across both the medium-lapse and long-lapse conditions, and there was no advantage for token-change over voice-change items. We interpret these results in terms of current models of lexical access in which episodic representation are available only immediately after presentation, while abstract representations are active for a longer time.

5aSC3. The relationship between speech perception and production: Evidence from children with speech production errors. Kathryn Cabbage (Commun. Sci. and Disord., MGH Inst. of Health Professions, 79 13th St., Boston, MA 02129, klcabbage@yahoo.com) and Thomas Carrell (Special Education and Commun. Disord., Univ. of Nebraska-Lincoln, Lincoln, NE)

Children with speech sound disorders (SSD), for reasons that are not well understood, fail to produce age-appropriate speech production targets in isolation, words, and/or functional conversation. One line of research suggests these children exhibit underlying perceptual deficits for acoustic cues that distinguish similar phonemes, exhibiting particular deficits for those phonemes produced in error (e.g., /r/, /l/) (Monnin and Huntington, 1974; Shuster, 1998), although other studies have reported mixed results in children with SSD (Locke, 1980; Rvachew and Jamieson, 1989). In the current study, children with SSD and typically developing children were presented with a perceptual discrimination task involving a phoneme produced in error by the SSD children (i.e., /r/). Findings revealed two separate subsets of children with SSD. The first exhibited poor discrimination of /r/ while a second exhibited exceptionally good discrimination of the same contrast. Given that all children with SSD exhibited distorted /r/ productions, a follow-up acoustic analysis is being conducted to identify specific acoustic characteristics of the distorted /r/s in the two groups (i.e., the poor /r/ discriminators versus the excellent /r/ discriminators). The pattern of results relating the acoustical details of /r/ production in these two populations should contribute to a more refined model of speech sound disorders.

5aSC4. Effects of talker sex on phonetic convergence in shadowed speech. Jennifer Pardo, Hannah Gash, Adelya Urmanche, Alexa Decker, Keagan Francis, Jaclyn Wiener, and Sara Parker (Psych., Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, pardo@optonline.net)

Many studies have reported phonetic convergence during speech shadowing and conversational interaction, with highly variable results. Some of this variability is likely due to effects of talker sex on phonetic convergence. For example, some studies only use a single male model, with both male and female shadowers, while others focus on only male or female talkers. To date, few studies have provided a rigorous investigation of variability across male and female talkers in phonetic convergence. Of these, some studies have found that women converged more than men, while others report the opposite pattern. The current study examined phonetic convergence in a speech shadowing task with 48 talkers (24 female) who shadowed same sex or opposite sex models. The results were analyzed to reveal effects of talker sex and item type (mono- versus bisyllabic words) on measures of phonetic convergence. Because phonetic convergence varies according to the sex of the talker and the model, future studies should employ talker sets that are balanced with respect to talker sex.

5aSC5. Interdependent processing of speech and background noise. Angela Cooper (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, akcooper@u.northwestern.edu), Susanne Brouwer (Behavioural Sci., Utrecht Univ., Utrecht, Netherlands), and Ann R. Bradlow (Linguist, Northwestern Univ., Evanston, IL)

Speech processing can often take place in listening conditions that involve the mixing of speech and background noise. This study used a speeded classification paradigm to investigate whether background noise is perceptually integrated with indexical (Exp. 1) and phonetic (Exp. 2) dimensions of the speech signal. In each experiment, English listeners classified words along one of two dimensions: noise (pure tone vs. white noise) or a speech dimension, either gender (male vs. female in Exp. 1a), talker identity (Sue vs. Carol in Exp. 1b), or phoneme (/p/ vs. /b/ in Exp. 2), while ignoring the other dimension, which could be held constant (control), co-vary (correlated), or vary randomly (orthogonal). The results indicated that background noise was not completely segregated from speech, even when the two auditory streams were spectrally non-overlapping. Perceptual interference was asymmetric, whereby irrelevant indexical and phonetic variation slowed noise classification to a greater extent than the reverse. This suggests that while context-specific information (e.g., noise) and within-signal speech features are coupled together, they are unevenly weighted during this early stage in processing. This asymmetry may stem from the fact that speech features have greater salience and are thus more difficult to selectively ignore than environmental noise.

5aSC6. Speaking style adaptations across the lifespan. Rachael C. Gilbert, Cristabella Trimble-Quiz, Karen Johnson, and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, 4812 Ave. H, Apt B, Austin, TX 78751, rachaelgilbert@gmail.com)

While intelligibility-enhancing speaking style modifications have been well researched in young adults, little is known about the development of speaking style adaptations across age. In particular, research on intelligibility-enhancing strategies in response to the environment (noise-adapted speech) and to the listener (clear speech) in children and older adult talkers is limited. In order to better understand how speech intelligibility varies with age, the present study examined conversational and clear speech sentences produced in quiet and in response to noise by 10 children (11–13 years old), 10 young adults (18–29 years old), and 10 older adults (60–84 years old). Twenty-two young adult listeners participated in word-recognition-in-noise and perceived age tasks. Results revealed that noise-adapted and clear speech modifications increased intelligibility for all talker groups compared to conversational speech produced in quiet, although children were marginally less intelligible. Perceived age strongly correlated with chronological age. Measures of F0, vowel space area, voice quality, spectral energy, and speaking rate were obtained for all talkers. Age-related differences were found for speaking rate, total energy in the 1-to-3 kHz region, and F0 mean. Further analyses will examine to what extent these acoustic-phonetic characteristics correlate with increases in intelligibility across speaking style adaptations and talker groups.

5aSC7. Talker's perspective modulates affective coherence in sentence comprehension. Hugo Quene, Anne R. Van Leeuwen, and Jos Van Berkum (Utrecht inst Linguist OTS, Utrecht Univ., Trans 10, Utrecht 3512JK, Netherlands, h.quene@uu.nl)

Does an audible frown or smile affect speech comprehension? Previous research suggests that a spoken word is recognized faster if its audible affect (frown or smile) matches its semantic valence. In the present study, listeners' task was to evaluate the valence of spoken affective sentences. Formants were raised or lowered using LPC to convey an audible smile or frown gesture co-produced with the stimulus speech. A crucial factor was the talker's perspective in the event being described verbally, in either first or third person. With first-person sentences, listeners may relate the talker's affective state (simulated by formant shift) to the valence of the utterance. For example, in "I have received a prize," a smiling articulation is congruent with the talker having experienced a happy event. However, with third-person sentences ("he has received a prize"), listeners cannot relate the talker's affective state to the described event. (In this example, the talker's affect can be empathic and positive, or envious and negative.) Listeners' response times confirm this hypothesized interaction: congruent utterances are processed faster than incongruent ones, but only for first-person sentences. When listeners evaluate spoken sentences, they combine audible affect, verbal content, as well as perspective, in a sophisticated manner.

5aSC8. Native listeners' sensitivity to foreign accent in short, slightly accented utterances: Non-native vowels. Hanyong Park (Linguist, Univ. of Wisconsin-Milwaukee, Curtin Hall 523, P.O. Box 413, 3243 N. Downer Ave., Milwaukee, WI 53211, park27@uwm.edu)

This study investigated whether native listeners can detect a foreign accent in short, slightly accented utterances. To answer this question, we examined 20 native listeners' sensitivity (d' values) to a foreign accent in a one-interval discrimination task (i.e., Yes-no design). Six L1 Korean learners of L2 English with high L2 proficiency along with six natives speakers of English produced the test materials consisting of three English vowels /ɑ/, /eI/, and /æ/ by using the delayed repetition technique. The listeners were asked to judge whether the speaker was a native or a non-native speaker of English for each speech sample played. Results indicated that most listeners detected a foreign accent from hearing the vowel stimuli. Furthermore, the listeners detected a foreign accent more often from the /æ/ stimuli than the /ɑ/ or the /eI/ stimuli. In line with previous L2 research, these results demonstrate that certain L2 segments are more difficult to learn than others. These results also suggest that listeners are sensitive to foreign accent and that they do not need much information (e.g., monosyllabic words) to detect a foreign accent, even in proficient L2 learners' productions.

5aSC9. Longer words or shorter words: Which one is more effective in distinguishing between self-identified gay and heterosexual male speakers? Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Prior research demonstrated that listeners relied primarily on vowels to determine a male speaker's sexual orientation (Tracy and Satariano, 2011). Additionally, listeners became more confident in their sexual orientation judgments if the utterance included more vowels (Tracy, 2013). Since multisyllabic utterances also contain more consonants, are listeners' judgments based on the number of vowels or consonants in an utterance? To investigate this question, monosyllabic, bisyllabic, and trisyllabic words were selected. Within each group of words, the number of phones varied. For example, a trisyllabic word contained either seven phones ("division") or nine phones ("contribute"). Listeners became more confident of the speaker's sexual orientation if the word included more vowels. With respect to the monosyllabic and bisyllabic items, listeners were less confident if the number of consonants in the word increased. Ratings were more confident for "have" compared to "help". The results for the trisyllabic words differed. Listeners became more confident if the word contained more consonants. "Contribute" resulted in more confident ratings than "division". The results demonstrated that listeners were more confident in the speaker's sexual orientation if the word included more vowels, but under certain conditions, confidence ratings did not improve if the utterance included additional consonants.

5aSC10. Emergent tonogenesis in Afrikaans. Andries W. Coetzee, Patrice S. Beddor (Dept. of Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan Str, Ann Arbor, MI 48109, coetzee@umich.edu), and Daan P. Wissing (Ctr. for Text Technol., North-West Univ., PotchefstRm., South Africa)

Afrikaans is usually described as having a contrast between prevoiced and voiceless unaspirated plosives ([b]-[p], [t]-[d]). This study documents an ongoing change in the pre-vocalic realization of the contrast. Preliminary data from five speakers show that all speakers produce some phonologically voiced plosives without prevoicing, with frequency of devoicing ranging from 30% to 85% across speakers. This devoicing is nearly categorical: VOTs of devoiced plosives average 12 ms and those of phonologically voiceless plosives average 17 ms. (By comparison, VOTs of voiced plosives average around Ő130 ms.) The contrast appears to be preserved in the F0 contour of the following vowel, which is 50 to 100 Hz lower (depending on the speaker) after phonologically voiced than after phonologically voiceless plosives. The F0 difference continues through at least 70% of the vowel.

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However, post-plosive F0 variation is not contingent on devoicing, e.g., F0 contours after phonological /b/ are the same regardless of whether the production is [p] or [b]. In these preliminary data, the magnitude of the F0 difference is linked to speaker age, with younger speakers showing a larger difference. Data from a larger group of speakers are being analyzed and will be presented.

5aSC11. Visualization of time-varying joint development of pitch and dynamics for speech emotion recognition. Chung Lee, Simon Lui (The Information Systems Technol. and Design Pillar, Singapore Univ. of Technol. and Design, 20 Dover Dr., Singapore 138682, Singapore, chung_lee@sutd.edu.sg), and Clifford So (School of Continuing and Professional Studies, Chinese Univ. of Hong Kong, Hong Kong, Hong Kong)

In this paper, we propose a new approach for visualizing the time-varying acoustic features for speech emotion recognition. Although the emotional state does not carry any linguistic information, it is a crucial factor that offers sentiment feedback to the listener. We propose to extract the two most prevalent acoustic features: pitch and dynamics, to identify the speech emotion of the speaker. We represent the time-varying pitch and dynamics as a trajectory in a two-dimensional feature space. Multiple trajectories are then segmented and clustered into signature patterns. This technique was successful in identifying and retargeting expressive musical performance styles. In evaluation, we use the German emotion language database. The database was created with ten professional actors (five males and five females) of ten emotionally unbiased sentences performed in six target emotions (Angry, Happy, Fear, Boredom, Sad, and Disgust). Results showed that the speech samples from the same actor of the same sentence but different emotions have dramatically different trajectory patterns. On the other hand, obvious common patterns were found among low valence emotions like Boredom and Sadness. The current study also opens future research opportunities for applying advanced pattern recognition techniques (e.g., Support Vector Machine and Neural Network) for better emotion identification.

5aSC12. Vowel discrimination at high fundamental frequencies in real speech. Daniel Friedrichs (Phonet. Lab., Univ. of Zurich, Plattenstrasse 54, Zurich 8004, Switzerland, daniel.friedrichs@uzh.ch), Dieter Maurer, Heidy Suter (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Zurich, Switzerland), and Volker Dellwo (Phonet. Lab., Univ. of Zurich, Zurich, Switzerland)

Previous research showed that, in singing, vowel qualities of isolated vowel sounds can be discriminated up to a fundamental frequency (F0) of about 500 Hz. However, indications are reported in literature for vowel discrimination on F0 > 500 Hz for singing (raised larynx condition, CVC context) as well as for speech-like sounds. In this study, we tested vowel discrimination at a high F0 in speech using minimal pairs build from eight long German vowels. Words were produced in speech mode at F0 of about 650 Hz by two female speakers. For all samples except the words including /a/ and /ɛ/, F0 exceeded F1 values as given in vowel statistics for Standard German. In a listening test, stimuli were played back in random order to 14 listeners (7f, 7m) for identification. The results showed that vowel discrimination can be preserved at such high fundamental frequencies. This could mean that, for our speakers and the high fundamental frequency examined, (1) source-filter-characteristics were effective up to 650 Hz, or (2) transitions played a crucial role, or (3) other spectral characteristics than formants have to be taken into account in order to explain these results.

5aSC13. Perceptual importance of time-domain features of the voice source. Marc Garellek (Linguist, Univ. of California, San Diego, La Jolla, CA), Gang Chen (Elec. Eng., Univ. of California, Los Angeles, Los Angeles, CA), Bruce R. Gerratt (Head and Neck Surgery, Univ. of California, Los Angeles, Los Angeles, CA), Abeer Alwan (Elec. Eng., Univ. of California, Los Angeles, Los Angeles, CA), and Jody E. Kreiman (Head and Neck Surgery, Univ. of California, Los Angeles, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu)

Our previous study examined the perceptual adequacy of different source models. We found that perceived similarity between modeled and natural voice samples was best predicted (in the time dimension) by the match between waveforms at the negative peak of the flow derivative (R^2 = 0.34). The extent of fit during the opening phase of the source pulses added only 2% to perceived match. However, in that study model, fitting was unweighted, and results might differ if another approach were used. In this study, we constrained the models to fit the negative peak of the flow derivative precisely. We fit 6 different source models to 40 natural voice sources, and then generated synthetic copies of the voices using each modeled source pulse, with all other synthesizer parameters held constant. We then conducted a visual sort-and-rate task in which listeners assessed the extent of perceived match between the original natural voice samples and each copy. Discussion will focus on the specific strengths and weaknesses of each modeling approach for characterizing differences in vocal quality, and on the importance of matches to specific time-domain events versus spectral features in determining voice quality. [Work supported by NIH/NIDCD grant DC01797 and NSF grant IIS-1018863.]

5aSC14. Using acoustic string-edit distance to evaluate US pronunciation variation. Clelia R. LaMonica (Dept. of English, Stockholm Univ., Stockholm 106 91, Sweden, clelia.lamonica@english.su.se)

String-edit distances, such as the Levenstien distance, have been used in perceptual-linguistic studies to compare differences among language varieties, based on changes that occur between two speech samples (Gooskens and Heeringa, 2004; Heeringa *et al*., 2006; Nerbonne *et al*., 2008). However, this method relies mainly on phonetic transcriptions rather than actual speech data. This work illustrates how acoustic measurements are taken from speech samples from across the United States, and a distance measurement between them is derived for use in further perceptual comparisons (such as perceived distance from standard, intelligibility, appeal, and identification). The speech samples consist of six sentences from various regions of the United States, each sentence containing phonological features that may be marked as perceptually significant for dialect identification (Clopper 2011, Labov *et al.* 2005, Thomas 2001). The methodology for assessing phonetic distance between two regional varieties is addressed, in particular, by using the Euclidean distance between normalized Bark formants of phonological features within the samples. The benefits and potential disadvantages to using acoustic data vs. transcribed data are addressed as well.

5aSC15. Mapping accent similarity and speech in noise intelligibility for British English accents. Paul Iverson, Melanie Pinet, and Bronwen G. Evans (Speech Hearing and Phonetic Sci., Univ. College London, 2 Wakefield St., London WC1N 1PF, United Kingdom, p.iverson@ucl.ac.uk)

Previous work has suggested that the recognition of speech in noise is affected by the accent similarity of the speaker and listener, as well as by the familiarity of the speaker's accent. The present study investigated this further by constructing multidimensional accent maps for British English speakers, as well as a small group of general American speakers, and examining how intelligibility within this space varied for listeners with different British English accents. The preliminary results suggest that speakers who have a central position in the accent space are most intelligible in noise, and that speakers with standard accents tend to occupy this central position. This implies that some aspects of speech intelligibility may be explained by the prototypicality of speakers within the broader accent space.

5aSC16. Lexically mediated perceptual learning generalizes to new word positions. Alexis R. Johns and James S. Magnuson (Univ. of Connecticut, 406 Babbidge Rd., Unit 1020, Storrs, CT 06269, alexis.johns@uconn.edu)

Acoustic properties of words vary based on idiolects of different speakers, yet listeners appear to understand this varying speech input effortlessly. How do listeners adapt to acoustic variation across speakers? Previous research has shown that listeners can implicitly shift phonetic category boundaries in a speaker-specific fashion during a two-step paradigm involving lexical decision on critical words with a word-medial ambiguous phoneme, followed by a phonetic categorization task of that ambiguous phoneme (see Kraljic and Samuel, 2005). Other studies report that this boundary shift does not generalize to other speakers (Kraljic *et al*., 2008).

However, for a given speaker, it does generalize to different positions within a word, for instance, from word-medial to word-initial (Jesse and McQueen, 2011). Importantly, these generalization tests used identical ambiguous phonemes in each position, rather than context-appropriate tokens, allowing the possibility of token-specific learning rather than true generalization. The current research extends these findings by providing distinct, position-appropriate tokens of the ambiguous phoneme during training and test items, and shows that the generalization to new word position is retained. Thus, generalization within a speaker is not token-specific, but instead exhibits some level of abstraction.

5aSC17. Perception of second language phoneme masked by first- or second-language speech in 20–60 years old listeners. Rieko Kubo, Masato Akagi (JAIST, 1-1 Asahidai, Nomi, Ishikawa 923-1292, Japan, rkubo@jaist.ac.jp), and Reiko Akahane-Yamada (ATR-IRC, Seika, Japan)

In previous research, we conducted a perceptual training on American English /r/-/l/ for Japanese adults in various age groups and found that; although the training improved identification of these phonemes in all age groups, the improvement decreased along the ages. Analyses suggested that perceptual cues of second-language (L2) phoneme perception differ across age groups. This paper investigated the decrease by focusing on perceptual cues. Listening tests were performed to assess intelligibility of Japanese words and English minimal-pairs masked by English or Japanese speech. Theoretically, in case L2 targets have phonologically non-equivalent in listener's first-language (L1) orthography, L1-relevant cues can be used separating the targets. In case L2 targets have phonologically equivalent in listener's L1 orthography, L2-relevant acoustic cues should be taken into perceptual cues. Masker having identical language to the perceptual cues may interfere identifying the targets. In the results of English targets which were supposed to be phonologically equivalent in Japanese orthography, young-adult Japanese showed higher effect from English masker than that of Japanese masker; conversely, older-adult Japanese showed higher effect from Japanese masker. The results suggested that young adults more focused on L2-relevant acoustic cues than elderly did. This difference may lead to the decrease in training effect.

5aSC18. Effect of aging on auditory processing: Relationship between speech perception and auditory brainstem responses. Su-Hyun Jin and Won So (Commun. Sci. and Disord., Univ. of Texas at Austin, 1 University Station, A1100, Austin, TX 78712, shjin@utexas.edu)

The purpose of the study is to evaluate the auditory processing abilities of older (60–75 years old) and younger (18–30 years old) listeners with normal hearing up to 4000 Hz. We are going to measure series of behavioral and physiological responses of these two listener groups. Behavioral measures include listener's hearing sensitivity, speech recognition in various listening conditions using different types of noise, and temporal resolution processing complex time-varying signals. For a physiological measure, speech-evoked ABRs, neural responses of subcortical auditory nuclei will be recorded. This quick and non-invasive technique allows us to measure neural responses to fast time-varying speech stimuli. We already synthesized speech stimuli, short syllable like /da/, /ba/, /ta/, /pa/ that can be used to evoke the ABR. It is hypothesized that compared to younger group, older listeners might show a loss of temporal precision results in abnormal ABRs such as subcortical timing delays and decreases in response consistency and magnitude. In addition, listeners with abnormal ABRs might show poorer performance in speech recognition in noise as well as reduced temporal resolution processing. That is, there would be significant correlation between behavioral and physiological measures.

5aSC19. Listen to your mother: Highly familiar voices facilitate perceptual segregation. Ingrid Johnsrude, Elizabeth Casey (Dept. of Psych., Queen's Univ., 62 Arch St., Kingston, ON K7L 3N6, Canada, ij4@queensu.ca), and Robert P. Carlyon (Cognition and Brain Sci. Unit, Medical Res. Council, Cambridge, United Kingdom)

We studied the effect of voice familiarity on the ability to segregate one voice from a competing speaker. Specifically, we examine the utility of arguably the most familiar voice of all—the mother's voice—in facilitating segregation, and compared it to the effect of a voice that listeners had been familiarized with in the laboratory. We tested 19 older adolescents (still living at home) on a version of the coordinate-response-measure procedure (CRM; Bolia *et al.*, 2001), with mixtures of two voices, at three signal-to-noise ratios (Ő3 dB, 0 dB, + 3 dB). Performance was better when the mother's voice was the target, compared both to novel and lab-familiar targets. At the most disadvantageous target-to-masker ratio (Ő3 dB), listeners were also better able to ignore their mother's voice so as to comprehend a stranger's voice more effectively, demonstrating that extremely familiar voice information facilitates segregation. This pattern of results is similar to that observed with older people (aged 44–59) when their spouse's voice was present in a two-voice CRM mixture (Johnsrude *et al.*, 2013). The new results demonstrate the importance of long-term (rather than short-term) familiarity and show that it aids sound segregation for adolescents as well as older adults.

5aSC20. Audio/visual correlates of a vowel near-merger in the Bay Area. Keith Johnson and Sarah Bakst (Linguist Dept., Univ. of California Berkeley, Berkeley, CA 94720, keithjohnson@berkeley.edu)

This study is a part of a research program aiming to learn the extent to which visual phonetic information may be involved in the maintenance of an acoustically weak or marginal contrast. The "caught/cot" contrast is one such contrast which has been merged in much of the western United States. We seek to document both the presence of the contrast in the SF Bay area and the phonetic realization of it. We will be reporting on audio/visual recordings from 33 native English speakers from the Bay Area (14 from San Francisco). Participants read a brief passage and a set of sentences that targeted these vowels, and then completed a commutation task where speakers tried to identify their own productions of the words "cot" and "caught." We found that three of the speakers reliably made the caught/cot contrast, and phonetic analysis confirms that regardless of whether speakers perceive a distinction, the vowels may differ acoustically or on degree of visual lip rounding, depending on the context. The two vowels tend to be most distinct in the single-word context and the least in the passage context. [Funded by the NSF.]

5aSC21. Acoustic cue weighting in perception and production of English and Spanish. Jessamyn L. Schertz, Natasha Warner, and Andrew J. Lotto (Univ. of Arizona, Douglass 200, Tucson, AZ 85721, jschertz@email.arizona.edu)

We present data quantifying the use of multiple acoustic dimensions in the perception and production of word-initial stops in Spanish and English in bilinguals and monolinguals, with a focus on the stability of individuals' cue weighting strategies across languages and modalities. Early Spanish-English bilinguals who use both languages on a regular basis, along with control groups of monolingual Spanish and English speakers, categorized stop-initial CV syllables covarying in Voice Onset Time (VOT), fundamental frequency at vowel onset (f0), first formant onset frequency (F1), and stop closure duration. Individual perceptual cue weights were calculated via logistic regression, and these weights were compared with the same speakers' use of the corresponding acoustic dimensions in their productions of the stop contrast. Preliminary results suggest that bilingual listeners' perceptual cue weighting strategies correspond across Spanish and English; that is, individuals who weight a given cue more heavily than average in English also weight that cue heavily in Spanish. However, in line with previous work, correspondences between a given individual's use of the acoustic dimensions across perception and production are more elusive.

5aSC22. Effects of auditory and visual variability on word learning in children. Kelly Casey, Shayna Marmon, Emily Butterworth, Elana Katz, Kristin A. Vasil-Dilaj, and Rachel M. Theodore (Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu)

One major feat in development is acquiring vocabulary for nouns, a process that necessitates learning a systematic pairing between an auditory and a visual stimulus. This is computationally complex because the acoustic signal produced for a given noun, such as "dog," varies considerably as do objects in the environment that are given this label. Studies that have

separately examined auditory and visual variability suggest that variability in either domain can facilitate word learning. The goal of the current work is to examine simultaneous contributions of auditory and visual variability on word learning. Typically developing infants (15–20 months of age) participate in nine weekly sessions. Each session entails exposure to nouns in a naturalistic play environment where children manipulate visual exemplars while listening to auditory exemplars. Auditory exemplars consist of productions by either ten talkers or by a single talker. Visual exemplars consist of objects that are highly variable or minimally variable in appearance. We measure learning for training exemplars, generalization to novel exemplars, and vocabulary size. Our research questions are (1) does variability in the auditory domain alone facilitate word learning as has been shown for visual variability, (2) to what degree does simultaneous variability in both modalities influences noun learning.

5aSC23. An auditory test battery for the assessment of executive functions. Blas Espinoza-Varas (Commun. Sci. & Disord., OU Health Sci. Ctr., 1200 N. Stonewall Ave., Oklahoma City, OK 73117-1215, blas-espinoza-varas@ouhsc.edu), Kai Ding, and Sudha Lakhwani (Biostatistics & Epidemiology, OU Health Sci. Ctr., Oklahoma City, OK)

Recent studies have examined the potential role of executive functions (EFs) on the speech communication ability of cochlear-implant, hearing-aid users, and other patients with disordered speech, language, or hearing; in many of the tests used to measure EFs, the stimuli are visual rather than auditory. We developed an Auditory Executive Function (AEF) test battery in which the stimuli consist of word commands (e.g., "quit," "stop") spoken either in stern or in lenient voice tone. The battery measures the speed and accuracy of voice-tone classification in four conditions: (1) in the absence of information or response conflict (baseline); (2) while attempting to inhibit potent but inappropriate responses prompted by conflicting ear-laterality information (inhibitory control); (3) while having to switch between incompatible response-mapping rules from trial to trial (cognitive flexibility); and (4) while having to monitor and update the word presented in the current trial and remember the word presented 2–3 trials prior to the current one (working memory). Relative to baseline, classification accuracy decreases with conflicting laterality information and more so with trial-to-trial switching of response-mapping rules, but only the latter condition decreased the classification speed. The AEF test battery is portable and fully automated. [Research supported by the Alcohol Beverage Medical Research Foundation.]

5aSC24. Perception of speaker sex in re-synthesized children's voices. Peter F. Assmann, Michelle R. Kapolowicz, David A. Massey (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, MS GR 41, Box 830688, Richardson, TX 75075, assmann@utdallas.edu), Santiago Barreda (Dept. of Physiol., Univ. of Arizona, Tucson, AZ), and Terrance M. Nearey (Dept. of Linguist, Univ. of AB, Edmonton, AB, Canada)

Recent studies have shown that fundamental frequency (F0) and average formant frequencies (FF) provide important cues for the perception of speaker sex. Experiments on vocoded adult voices have indicated that upward scaling of F0 and FFs increases the probability that a voice will be perceived as female while downward scaling increases the probability that the voice will be perceived as male. The present study extends these manipulations to children's voices. Separate groups of adult listeners heard vocoded /hVd/ syllables spoken by five boys and five girls from 14 age groups (5–18 years) in four synthesis conditions using the STRAIGHT vocoder. These conditions involved swapping F0 and/or FFs to the opposite-sex average within each age group. Compared to the synthesized, unswapped originals, both the F0-swapped condition and the FF-swapped condition resulted in lower sex recognition accuracy for the older females but relatively smaller effects for males. The combined F0 + FF swapped condition produced the largest drop in performance for both sexes, consistent with findings indicating that a change in F0 or FFs alone is generally insufficient to produce a compelling conversion of speaker sex in adults. [Hillenbrand and Clark, Attention Percep. Psychophys. **71**(5), 1150–1166 (2009).]

5aSC25. Voice quality variation across gender identities. Kara Becker, Sameer ud Dowla Khan, and Lal Zimman (Linguist, Reed College, 3203 SE Woodstock Bvld., Portland, OR 97202, sameeruddowlakhan@gmail.com)

While several studies of the phonetics of American English make reference to general features of "women's speech" and "men's speech," no large-scale acoustic or articulatory study has established the actual range and diversity of phonetic variation along gender identities, encompassing different sexual orientations, regional backgrounds, and socioeconomic classes. The current study explores the acoustic and articulatory phonetic features related to gender, particularly focusing on voice quality, e.g., creaky and breathy phonation. Subjects identifying with a range of gender and other demographic identities were audio recorded producing the Rainbow Passage as well as wordlists targeting variation in vowel quality. Simultaneously, electroglottographic (EGG) readings were taken and analyzed for voice quality. Subjects were then asked to rate recordings of other people's voices to identify the personal characteristics associated with the acoustic reflexes of phonation; in the final task, subjects were also explicitly asked about their language ideologies as they relate to gender. Thus, the current study explores the relation between gender identity and phonetic features, measured acoustically, articulatorily, and perceptually. This work is currently underway and preliminary results are being compiled at this time.

5aSC26. The role of acoustics in processing syntactically ambiguous sentences. Victoria Sharpe (Honors College, Univ. of South Carolina, 104 Mount Elon Church Rd., Hopkins, SC 29061, sharpevp@email.sc.edu), Dirk-Bart den Ouden, and Daniel Fogerty (Dept. of Commun. Sci. and Disord., Univ. of South Carolina, Columbia, SC)

While natural speech prosody facilitates sentence processing, unnatural prosody is expected to decrease speed and accuracy in resolving syntactic ambiguities. This study investigated how, and to what extent, fundamental frequency (F0) and temporal envelope (E) contribute to sentence processing. Signal processing methods degraded either F0 or E in 120 garden-path sentences with a temporarily ambiguous structure (e.g., While the man hunted the deer ran into the woods). Participants listened to natural and acoustically modified garden-paths and, for each, answered a comprehension question and repeated the sentence. Results demonstrated that degrading E consistently affected sentence comprehension, with a different effect observed for degrading F0. Interestingly, the effect of acoustic degradation interacted with effects of verb type and sentence plausibility. In the E-modified condition, sentences with Reflexive Absolute Transitive verbs in which the erroneous interpretation was plausible (e.g., While Mary bathed the baby fell in the tub—*Mary bathed the baby) resulted in productions that were relatively shorter in duration and that reconstructed the original sentence prosody. These findings suggest E plays a greater role, compared to F0, in disambiguating syntax. However, use of prosody by listeners may vary based on context, suggesting prosodic information can interact with cognitive processing load.

5aSC27. Acoustic separability of Mandarin tonal categories in careful and conversational speech. Daniel Brenner (Linguist, Univ. of Arizona, 814 E 9th St. Apt. 14, Tucson, AZ 85719-5322, dbrenner@email.arizona.edu)

Previous work has found that tonal categories in conversation are realized with mostly much weaker acoustic cues than in careful speech. For example, pitch curves become flatter (less curved), more level (shallower slopes), and more centralized (closer to the speaker's mean speaking pitch) in conversation on average relative to careful speech, and also more variable. At the same time, a few cues appear to change direction (e.g., the pitch range of tone 4 is much larger than that of tone 3 in careful speech, but is far shorter in conversation), or become acoustically salient to categories in conversation only (pilot results suggest that tone 3 is especially short in duration in conversation). The present project incorporates tonal context, and bigram syllable- and tone- frequencies into a model of the differentiability of second-syllable tonal categories in two-syllable Mandarin words in conversation. This model approximates the extent to which tonal category can be determined from available acoustic cues in conversation while accounting for co-occurrence frequencies.

5aSC28. Production and perception of the pre-lateral, non-low, back vowel merger in northeast Ohio. Lacey R. Arnold (English, North Carolina State Univ., 1214 Carlton Ave., Raleigh, NC 27606, lrarnold@ncsu.edu)

This study examines the status of the pool/pull, pole/pull, and pool/pull/pole mergers in the Youngstown, Ohio area, as well as various factors that may influence perception of these mergers. The participants include 16 natives of the Youngstown, Ohio area, ranging from ages 9–66, and making up two separate families; one individual belonged to neither. The findings suggest that all three mergers can be found in the Youngstown area, though each is undergoing a different process. Results of the perception task suggest that individuals can use additional cues, such as vowel duration, to aid in identification of words containing merged phonemes. Findings also suggest that homophonous words uttered by merged speakers in the context of a carrier sentence may contain additional information that listeners are able to access and utilize for word identification. Lastly, the effects of familiarity with the speaker and merged/distinct production on word identification reveal no definitive results, though the study's findings suggest further investigation of these factors would be a worthwhile endeavor.

5aSC29. Perceptual tonal spaces in whispered speech. Grace Kuo (Linguist, Macalester College, 3125 Campbell Hall, Los Angeles, CA 90095, gracekuo@humnet.ucla.edu)

Whispered speech is made when no vocal fold vibration occurs. However, pitch from whispers can still be perceived for some listeners because it corresponds to the change in F1 and F2. Listeners rely on the laryngeal movements, which not only associate with pitch change but also result in the change of the tongue position as well as the shape of the oral cavity. This study examines the acoustical-perceptual relationships in the identification of pitch during whispered tone production. Listeners (English and Taiwanese) rate the similarity of fifty AX tonal pairs (Taiwanese [kaʊ̆] and [ç] in modal voice vs. whispered voice). Multidimensional scaling (MDS) analysis is used to map English listeners' and Taiwanese listeners' perceptual tonal spaces regarding modal speech and whispered speech stimuli. The correlations between the similarity ratings, the reciprocal of their reaction time, and the selected acoustic measures are examined.

5aSC30. Dynamic pitch and pitch range interact in distortions of perceived duration of American English speech tokens. Alejna Brugos and Jonathan Barnes (Boston Univ., 14 Asylum St., Mendon, MA 01756, abrugos@bu.edu)

Previous research showed that pitch factors can distort perceived duration: tokens with dynamic or higher f0 tend to be perceived as longer than comparable level-f0 or lower-f0 tokens, and silent intervals bounded by tokens of widely differing pitch are heard as longer than those bounded by tokens closer in pitch (the kappa effect). Fourteen subjects were asked to judge which of two exemplars of a spoken word sounded longer. All tokens were created from the same base file with manipulations of objective duration, f0 contour (plateaux vs. rises of different slopes) and pitch range. Results show that pitch range relation between the two exemplars was a stronger predictor of perceived duration distortion than f0 contour. In addition to previously demonstrated effects of f0 height (Yu, 2010), greater f0 discontinuity between tokens increases the likelihood that the first token of a pair will be judged as longer, suggesting that some previous findings showing the effects of dynamic pitch on perceived duration may actually be magnified by the kappa effect. Listeners may be responding to perceived prosodic distance that integrates information from timing (filled and silent intervals) and pitch (pitch slope and pitch jumps across silent intervals).

5aSC31. Indexical variation affects semantic spread. Ed King and Meghan Sumner (Linguist, Stanford Univ., 450 Serra Mall, Bldg. 460, Rm. 127, Stanford, CA 94305, etking@stanford.edu)

The role of indexical variation in spoken word recognition is constrained to acoustically rich lexical representations. Theoretically, lexical activation depends on indexical variation, but subsequent processes like associative semantic spread depend on activation strength, not indexical variation. Social psychological theories view indexical variation as integral to online processes such as persona construal. Therefore, information gleaned from indexical variation might pervade spoken word recognition more broadly. We investigate the effects of indexical variation on semantic activation in word-association and semantic-priming paradigms. Across three studies, we show that top associate responses depend on the voice of the probe word ("space" in man's voice: time; woman's voice: star; child's voice: planet). Voice also affects response frequency distributions: the man's voice receives a wider variety of weaker responses, while the woman's and child's voices receive fewer, stronger, responses. We also find that semantic priming varies as a function of voice-specific word association strength: priming is stronger to strong voice-specific associates (woman: space-star) than to weak associates (woman: space-time). We argue that indexical variation affects spoken word recognition beyond an episodic lexicon and provide an account capturing effects of learned associations between acoustic patterns and linguistic and social features in spoken language processing.

5aSC32. Effects of listener characteristics on foreign-accentedness rating of a non-standard English dialect. Andrea Morales and Natasha Warner (Linguist, The Univ. of Arizona, 5242 S Hampton Roads Dr., Tucson, AZ 85756, andreamorales@email.arizona.edu)

This project analyzes what characteristics of listeners affect whether they perceive Chicano English as foreign-accented English. Many Americans assume Chicano English (CE) is non-native English spoken by native Spanish speakers, but CE is often spoken as a native dialect of English. CE is a very common dialect in Tucson, Arizona, and this project examines the correlation between listeners' ethnicity, familiarity with Hispanic people, and political stance on immigration, and their perception of CE as foreign-accented. Stimuli are sentences read by CE and other Tucson speakers that contain phonetic environments where CE has features that distinguish it from Standard American English (SAE) . The listener population is Southern Arizonans of various ethnicities with varying degrees of exposure to CE and Spanish. The experiment uses a Foreign Accentedness Rating (FAR) task, as well as classification of stimuli as spoken by a Hispanic vs. Anglo speaker and background questions on listeners' language background and political opinions. Highly accurate identification of ethnicity is predicted, as well as correlations between some measures of the listeners' background and strength of FAR rating of CE speakers. Conclusions involve the effect of long-term exposure to a local dialect and sociolinguistic status on perceived degree of foreign accent.

Session 5aSP

Signal Processing in Acoustics: Signal Processing Models for Sound Production and Perception

Mark D. Skowronski, Chair
Communicative Sciences and Disorders, Michigan State Univ., Okemos, MI 48864

Contributed Papers

8:30

5aSP1. Distribution and propagation of sound sources in the vocal tract. Liran Oren, Sid Khosla, and Ephraim Gutmark (Univ. of Cincinati, PO Box 670528, Cincinnati, OH 45267, orenl@ucmail.uc.edu)

In the current study, an array of 128 microphones is strategically positioned outside a vocal tract model that is placed above an excised canine larynx. Acoustic holography technique is then used to identify the distribution of sound in the vocal tract by using simultaneous measurements from the microphone array. The results show that with no downstream constriction (i.e., open mouth) the energy coming from the higher harmonics is concentrated near the vibrating folds. When downstream constriction is added (i.e., partially closed mouth), the energy in the higher harmonics is shifted further downstream toward the mouth. These results suggest that acoustic holography can be used as a new tool to assess how sound sources propagate in the vocal tract during normal speech and consequently determine the severity of certain speech disorders such as hypernasality and nasal emission.

8:45

5aSP2. Accuracy of six techniques for measuring formants in isolated words. Christine H. Shadle, Hosung Nam (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu), and D. H. Whalen (Speech-Language-Hearing Sci., City Univ. of New York, New York, NY)

In previous work comparing formant measurements of synthetic vowels, manual measurements and four of the six algorithms tested were shown to have large errors and be biased by fundamental frequency (F0). In this study, natural speech recorded by five speakers (2 female, 3 male) was used to provide a more realistic test. The context /hVd/ was used with vowels V = {/i, ae, u, a/}, in which formant frequencies could be expected to be relatively constant; declarative intonation was elicited so that F0 would be falling throughout the vowel, eliciting evidence of F0 bias. Formants were estimated throughout using LPC Burg, LP closed-phase covariance, Weighted Linear Prediction with Attenuated Main Excitation (WLP-AME) [Alku *et al.*, JASA **134**(2), 1295–1313 (2013)], AVG, CEPS, and reassigned spectrogram with pruning (RS). The LPC methods show evidence of F0 bias, with WLP-AME having the smallest errors (Alku *et al.* compared it to many LPC variants, including DAP); RS has the smallest errors overall, but is not fully automatic. Accuracy during vocal fry and for low vs. high F0s will be discussed. As with synthetic speech, WLP-AME can be recommended as the analysis of choice until RS becomes more automatic and less subjective. [Work supported by NIH.]

9:00

5aSP3. Auditory-inspired pitch extraction using a synchrony capture filterbank for speech signals. Kumaresan Ramdas, Vijay Kumar Peddinti (Dept. of Elec., Comput. & Biomedical Eng., Univ. of Rhode Island, 4 East Alumni Ave., Kingston, RI 02881, kumar@ele.uri.edu), and Peter Cariani (Hearing Res. Ctr. & Dept. of Biomedical Eng., Boston Univ., Boston, MA)

The question of how harmonic sounds produce strong, low pitches at their fundamental frequencies, F0s, has been of theoretical and practical interest to scientists and engineers for many decades. Currently the best auditory models for F0 pitch, [e.g., Meddis and Hewitt, J. Acoust. Soc. Am. **89**(6), 2866–2894 (1991)] are based on bandpass filtering (cochlear mechanics), half-wave rectification and low-pass filtering (haircell transduction and synaptic transmission), channel autocorrelations (all-order interspike interval statistics) aggregated into a summary autocorrelation, and an analysis that determines the most prevalent interspike intervals. As a possible alternative to autocorrelation computations, we propose an alternative model that uses an adaptive Synchrony Capture Filterbank (SCFB) in which groups of filter channels in a spectral neighborhood are driven exclusively (captured) by dominant frequency components that are closest to them. The channel outputs (for frequencies below 1500 Hz) are then adaptively phase aligned with respect to a common time reference to compute a Summary Phase Aligned Function (SPAF), aggregated across all channels, from which F0 is easily extracted.

9:15

5aSP4. Acoustic characteristics of the Lombard effect from talkers with Parkinson's disease. Rahul Shrivastav (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), Mark D. Skowronski (Communicative Sci. and Disord., Michigan State Univ., 2355 Club Meridian Dr., Apt. B02, Okemos, MI 48864, markskow@hotmail.com), Lisa M. Kopf, and Brad Rakerd (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Speech production in noise elicits the Lombard effect, characterized by an increase in vocal effort. In talkers with Parkinson's disease (PD), noisy environments are a stressor to speech production. Recently, the intelligibility of speech from PD talkers speaking in noisy (additive multi-talker babble) and reverberant environments was shown to be more sensitive to both factors than speech from healthy talkers [Kopf *et al.*, ASHA **7387** (2013)]. The current study investigated the acoustic properties of speech and how they change in noise and reverberation for PD talkers compared to healthy talkers. Healthy talkers increased their level and fundamental frequency in additive noisy environments and, to a lesser degree, in reverberant environments. PD talkers also changed their level and fundamental frequency but to a more limited extent compared to healthy talkers. Additional changes in speech production (speaking rate, articulation range, and voice quality) were observed in individual talkers from both groups.

9:30

5aSP5. The cepstral peak: A theoretic analysis and implementation comparison of a popular voice measure. Mark D. Skowronski, Rahul Shrivastav, and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Rm. 216, East Lansing, MI 48824, markskow@msu.edu)

Cepstral analysis of speech has traditionally provided two pieces of information: (1) the cepstral lag, which is a measure of fundamental frequency, and (2) the cepstral peak, which is a measure of the degree of periodicity. Cepstral analysis conveniently separates the effects of the vocal tract from the acoustic source without explicit vocal tract estimation. The cepstral peak of speech is a popular diagnostic tool for voice disorders and vocal treatment assessment, yet the expected value of the cepstral peak has not received sufficient theoretic treatment. Discrete-time analysis of the

cepstrum of a periodic pulse train revealed that the cepstral peak is 1/2 for all pulse trains with integer fundamental period T samples. For the more general case of a non-integer period, a pulse train formed with sinc functions produces a cepstral peak of $1/2+\varepsilon$ where the magnitude of ε scales with 1/T. The cepstral peak of various test signals was compared to cepstral peak prominence [Hillenbrand *et al*., J. Speech Hear. Res. **37**, 769–778 (1994)], and accuracy was improved by (1) zero-padding the log spectrum before inverse Fourier transformation, which provides cepstral interpolation, and (2) limiting spectral nulls, which trades off estimate variance and bias.

9:45

5aSP6. Objective measures of blind source separation and the intelligibility of separated speech. Richard Goldhor, Keith Gilbert (Speech Technol. & Appl. Res. Corp., 54 Middlesex Turnpike, Entrance D, Bedford, MA 01730, rgoldhor@sprynet.com), Suzanne Boyce, and Sarah Hamilton (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

When multiple acoustic sources are present in an environment monitored by multiple microphones, each microphone's response signal is typically a weighted and delayed mixture of the active source signals. Subject to important constraints, these mixtures can be processed using Blind Source Separation (BSS) methods to construct output signals representative of each of the "hidden" acoustic sources in isolation. An important objective measure of the quality of a given "separation solution" is the amount of mutual information remaining in the BSS output channels: the less mutual information across channels, the better the separation. Often, the acoustic signals of interest in a complex acoustic environment are speech sources, and the measure of separation quality that really matters to listeners is speech intelligibility—a subjective (perceptual) measure that cannot be directly computed by a signal processing system. Thus, it would be useful to know how well objective measures of separation quality, such as mutual information, correlate with intelligibility. We present results of perceptual and objective tests exploring the relationship between objective separation metrics and word intelligibility as estimated by listeners' responses in word identification tasks, for target-word-in-carrier-phrase utterances recorded in an acoustic environment with masking speech babble and broadband noise sources.

10:00–10:15 Break

10:15

5aSP7. Toward a scalable, binaural hearing device: Acoustics, interaural processing, and scene decomposition. Birger Kollmeier, Marko Hiipakka, Giso Grimm, Tobias Neher, Tobias de Taillez (Cluster of Excellence Hearing4all & Medizinische Physik, Universität Oldenburg, Oldenburg D-26111, Germany, birger.kollmeier@uni-oldenburg.de), Jens Schröder (Project Group Hearing, Speech and Audio Technol., Fraunhofer IDMT, Oldenburg, Germany), Jörn Anemüller, and Volker Hohmann (Cluster of Excellence Hearing4all & Medizinische Physik, Universität Oldenburg, Oldenburg, Germany)

One of the aims of the focused research group "Individualized hearing acoustics" are elements of an assistive listening device that both fits the requirements of near-to-normal listeners (i.e., providing benefit in noisy situations or other daily life acoustical challenges) and, in addition, can be scaled up to a complete hearing aid for a more substantial hearing loss. A review is given on the current work on acoustically "transparent" open-fitting earpieces, algorithms for interaural cue enhancement, and for automatically detecting relevant changes in the acoustical scene that call for an altered processing strategy. The current prototype runs on the binaural, cable-connected master hearing aid (MHA) that includes earpieces that allow for approaching acoustic transparency. This can be achieved by individual calibration of the earpieces using in-situ in-ear measurements and electro-acoustic models of the earpieces and the ear canal. A binaural high-fidelity enhancement algorithm motivated by interaural magnification is evaluated in its benefit for normal and near-to-normal listeners. Finally, several acoustic event detection schemes are compared with respect to their applicability in real-life situations. They decompose the ambient acoustic signal into spectro-temporal basis functions and utilize machine learning classifiers to detect typical natural and artificial events. [Work supported by DFG.]

10:30

5aSP8. The effects of head-worn attire on measured head-related transfer functions. Stephen Nimick (Beats Electronics, LLC, 3201 Sawtelle Blvd., Apt. 108, Los Angeles, CA 90066, stephen.nimick@beatsbydre.com), Joshua Atkins, and Ismael Nawfal (Beats Electronics, LLC, Santa Monica, CA)

In this paper, we investigate measured changes in spatial perception presented by hats and other head-worn equipment. Particularly, helmets worn in sports, military and civil services (firefighters, police etc.) can affect spatial awareness and localization of wearers, possibly putting the wearer and others in danger. For this paper, a variety of hats were placed on a Head And Torso Simulator (HATS). The Head Related Transfer Function (HRTF) was measured in an anechoic chamber at a variety of angles around the HATS. The data will be analyzed to show the differences between various head-worn devices.

10:45

5aSP9. The effects of spectral fine structure and amplitude envelope on perceived similarity and identification of approaching vehicles. Jeremy Gaston (Army Res. Lab., Aberdeen Proving Grounds, Aberdeen, MD, jeremy.r.gaston.civ@mail.mil), Kelly Dickerson (Army Res. Lab., Abingdon, MD), and Mark Ericson (Army Res. Lab, Aberdeen, MD)

The recognition and identification of important complex natural sound sources is a critical component of maintaining environmental awareness. The properties of sound sources are not static; environmental factors can cause dramatic shifts in acoustic structure. The present study investigates the effects of large imposed acoustic changes on listener perception of approaching ground and aerial vehicles. In the first experiment, listeners rated the perceived similarity of the vehicle stimuli that were either (1) unaltered, or had (2) spectral fine structure replaced with random noise, or (3) had their natural amplitude envelopes flattened. The results of a multidimensional scaling (MDS) analysis found that the ordering of perceptual space for each of the conditions was generally a function of the acoustic information available. The second experiment investigated listener identification for the same sets of sounds. Identification performance across vehicle sounds had good agreement with similarity mappings and was best when spectral fine structure was smoothed and variations in amplitude envelope were maintained, and worst when the opposite was true. These results are consistent with the notion that amplitude envelope is a critical component in the recognition and identification of complex natural sound sources.

11:00

5aSP10. Assessment of heart rate variability from speech analysis. Nivedita Deshpande (School of Studies in Electronics & Photonics, Pt. Ravishankar Shukla Univ., Raipur, India), Kavita Thakur (School of Studies in Electronics & Photonics, Pt. Ravishankar Shukla Univ.,Raipur, Chhattisgarh 492010, India, kavithakur@rediffmail.com), and Arun S. Zadgaonkar (Electronics and TeleCommun. Eng., Dr C.V.Raman Univ., Bilaspur, Chhattisgarh, India)

In this paper, various methods of heart beat variability assessment from speech analysis have been presented. Heart rate variability (HRV) is a physiological phenomenon where the time interval between heart beats varies. It is measured by the variation in the beat-to-beat interval. The present work deals with the HRV detection from speech parameters. RR-cycle detects one heart beat. Continuous monitoring of electrocardiograph for a span of time can provide HRV. More than 250 samples of normal informants as well as heart patients with enlarged heart have been collected during the four years of research work. The regular ECG and speech samples of the patient have been collected and analyzed. Further they had compared with the parameters of a normal healthy informant. Speech samples were collected through a microphone and subjected to be digitized. The required speech segmental have been extracted and analyzed through a DSP tool, PRAAT. ECG sample has been recorded through an ECG machine. A technique of HRV detection from speech analysis has been presented in this paper. The HRV detection from speech can be a very helpful tool in monitoring the functioning of human heart.

11:15

5aSP11. Temporal stability of long-term measures of fundamental frequency. Pablo Arantes (Universidade Federal de São Carlos, Via Washington Luís, Km. 235 - Caixa Postal 676, São Carlos 13565-905, Brazil, pabloarantes@gmail.com) and Anders Eriksson (Dept. of Philosophy, Linguist and Theory of Sci., Univ. of Gothenburg, Gothenburg, Sweden)

We investigated long-term mean, median, and base value of the voice fundamental frequency (F0) to estimate how long it takes their variability to stabilize. That information can be useful in the development of F0 contour normalization and forensic applications. Change point analysis was used to locate changes in underlying variance in the mean, median, and base value time-series. In one experiment, stabilization points were calculated in recordings of the same text spoken in 26 languages. Average stabilization points are 5 s for base value and 10 s for mean and median. Variance after the stabilization point was reduced around 40 times for mean and median and more than 100 times for the base value. In another experiment, four speakers read two different texts each. Stabilization points for the same speaker across the texts do not exactly coincide as would be ideally expected. Average point dislocation is 2.5 s for the base value, 3.4 for the median, and 9.5 for the mean. After stabilization, differences in the three measures obtained from the two texts are 2% on average across speakers. Present results show that stabilization points in long-term measures of F0 occur earlier than suggested in the previous literature.

11:30

5aSP12. Improvement of speech coding algorithm based on DSP system. Xiaochen Wu (Harbin Eng. Univ., 803, Shushing Bldg., Harbin 150000, China, wuxiaochen629@163.com) and HaiNing Lv (Harbin Eng. Univ., Harbin, Heilongjiang Province, China)

With the rapid development of the communication technology, the high quality of speech communication has become one of the main development trends. In order to solve the quality problem in the processing of communication, pre emphasis is used to improve the quality of the high frequency components in codec algorithm. The speech enhancement of spectral subtraction is also applied in this speech coding algorithm to improve the ability of anti noise. Repackaging the sending frames and redefining the length of the frame that need to be send in the algorithm in order to realize the low speed in speech communication process. At last, the speech quality will be proved by PESQ—MOS test and the spectrogram. From the test and the figure, the quality of the synthesized speech especially with background noise has been significantly improved.

11:45

5aSP13. Acoustic transfer functions and speech intelligibility in permanent wearing headsets. Vladimir G. Popov (Lab. of Phys. of Semiconductor Nanostructures, Inst. of Microelectronics Technol. of Russian Acad. of Sci., Academician Osipyan St. 6, Chernogolovka 142432, Russian Federation, popov@iptm.ru), Alexey L. Ushakov (Necktec, Moscow, Russian Federation), Zintars Laris, Serge Batov, and Yurii Saprovskii (R&D Akustika, Riga, Latvia)

A detailed analysis of the acoustic transfer functions has been investigated for different locations of the headset microphones in the area of human head. Amplitude-frequency and phase-frequency characteristics of acoustic signals are discussed in the report. The signals have been recorded with the micro-mechanical-electrical-system (MEMS) microphones located on a mannequin, simulating the acoustic properties of the human torso. The the source of the signals is an emitter built in the mouth of the manikin's head. Analysis of the characteristics has shown that the acoustic signals undergo significant phase and amplitude changes associated with the interference and diffraction processes. These changes result in significant nonlinearity phase characteristics, and this nonlinearity depends on the installation location of the MEMS microphone. Also quality of speech intelligibility has been estimated by the method of expert evaluation and the speech has been recorded for different positions of the microphones and speaker head. A comparative analysis of the results has shown the optimal location of the microphones in the ear and on the breast of man.

FRIDAY MORNING, 9 MAY 2014 — 556 A/B, 8:00 A.M. TO 12:00 NOON

Session 5aUW

Underwater Acoustics: Underwater Acoustic Propagation

Ilya A. Udovydchenkov, Chair
Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., MS #9, Woods Hole, MA 02543

Contributed Papers

8:00

5aUW1. Modeling three-dimensional propagation in a continental shelf environment using the finite element method. Benjamin Goldsberry and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bmgoldsberry@gmail.com)

Horizontal refraction is a well known three-dimensional propagation effect in underwater continental shelf environments. However, only a few experimental observations of this phenomenon exist. In 2007, horizontal refraction was not only experimentally verified off the coast of Florida but was found to dominate the acoustic propagation. Numerical models that allow for azimuthal variations in the geometry are needed to accurately model this environment. In this work, a three-dimensional finite element method with the longitudinally invariant technique is presented. In this method, a cosine transform is applied to the three-dimensional Helmholtz equation to remove the range-independent dimension. The finite element method is used to solve the transformed Helmholtz equation for each out-of-plane wavenumber. The inverse cosine integral is then used to transform the pressure field back to three-dimensional spatial coordinates. The computed pressure field is then suitable for comparison to experimental measurements. [Work supported by ONR, Ocean Acoustics.]

8:15

5aUW2. Explanation of an extended arrival structure measured in the Catoche Tongue using a three-dimensional propagation model. Megan S. Ballard, Jason D. Sagers, and David P. Knobles (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

This paper presents data recorded on a sparsely populated vertical line array (VLA) moored in the center of the Catoche Tongue, a major reentrant in the Campeche Bank in the southeastern Gulf of Mexico. An extended arrival structure observed in the pressure time series data resulting from signals underwater sound (SUS) deployed 50 to 80 km from the VLA is examined. Two distinct classes of arrivals are associated with the reception of each SUS event. The first class of arrivals was identified as resulting from the direct path, and the second class of arrivals is believed to result from horizontal refraction from the margins of the tongue. The difference in time between the onset of the first and second class of arrivals decreases as the distance from the VLA increases in a manner that is consistent with a decrease in the length of the refracted path relative to the direct path as suggested by the environment geometry. The observations are further supported by a three-dimensional (3D) acoustic propagation computation which reproduces many of the features of the measured data and provides additional insight into the details of the 3D propagation. [Work supported by ONR.]

8:30

5aUW3. Numerical modeling of three-dimensional underwater sound propagation under rough sea surfaces. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

A three-dimensional parabolic-equation (PE) solution of long-distance underwater sound propagation with the influence of arbitrary sea surface roughness is derived. A stair-step approximation is implemented to the sea surface height along the PE solution marching direction. A higher-order operator splitting algorithm is utilized for an Alternative Direction Implicit (ADI) method to achieve computational efficiency and accuracy. The ADI method also ensures that the pressure release interface condition on the irregular sea surface is satisfied exactly. An theoretical solution will be used to benchmark the PE solution, and examples of sound propagation under sea surface wave swells will be shown. Depending on the source position relative to the waves, the acoustic waveguide condition can change from ducting to anti-ducting. Two-way PE solutions to include acoustic backscattering from rough sea surfaces will also be discussed. [Work supported by the ONR.]

8:45

5aUW4. Horizontal refraction and whispering gallery sound waves in area of curvilinear coastal wedge in shallow water. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, Mt Carmel, Haifa 31905, Israel, katz@phys.vsu.ru) and Andrey Malykhin (Phys. Dept, Voronezh Univ., Voronezh, Russian Federation)

Horizontal refraction of sound signals in area of coastal wedge determines specific structure of the sound field in horizontal plane, including multipath areas and caustics (in the framework of approximation of horizontal rays and vertical modes). This structure depends on mode number and frequency. In the paper sound field of signal, radiated by the point source is studied in the shallow water area which can be considered as a wedge with curvilinear coastal line (lake, gulf, cape, etc.). It is shown that for some distances from the source to the beach whispering gallery waves (whispering gallery horizontal rays) exist, propagating along the coast, where energy is concentrated in narrow area in horizontal plane. Structure and characteristic of these rays depend on mode number, frequency and waveguide parameters (in particular, steepness of sea floor and radius of curvature). Intensity fluctuations as well as variation of pulse shape and frequency spectrum are studied; analytical estimations are presented as well as discussion of possible experimental observations. [Work was supported by RFBR and BSF.]

9:00

5aUW5. The effect of large- and small-scale sound velocity field structure on shallow water sound propagation in the East China Sea. Clare G. Nadig (Graduate Program in Acoust., The Penn. State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, cgn5004@psu.edu) and David L. Bradley (Appl. Res. Lab., The Penn. State Univ., State College, PA)

During the Transverse Acoustic Variability Experiment (TAVEX) in 2008, towed CTD measurements were used to record the sound speed field over an area of the East China Sea surrounding 34- and 20-km acoustic paths. The data revealed several propagating internal wave packets along with sound speed variability due to density-compensated thermohaline variations (spice). Concurrent acoustic data were also recorded from 300- and 500-Hz sources. Parabolic equation modeling is used to examine the effects of the measured internal waves and spice on acoustic propagation and to make comparisons with the recorded acoustic data.

9:15

5aUW6. Relationships between acoustic surface wave shape inversion and receiver array statistics in a variety of sea states. Sean Walstead and Grant Deane (ECE/SIO, UCSD, 9500 Gilman Dr., 0407, La Jolla, CA 92093-0407, swalstead@ucsd.edu)

Mid-frequency (12 kHz) acoustic scattering from the sea surface is analyzed with regard to surface wave shape inversion and receiver array characteristics. The forward scattered data is from the Surface Processes and Communications Experiment (SPACE08). Multipath arrivals representing surface, bottom-surface, and surface-bottom paths are distinguishable, implying that knowledge of the surface is known approximately 1/3, 1/2, and 2/3 the distance between source and receiver. Surface wave shape inversion at those ranges are presented in a variety of environmental conditions and compared to other calculable statistics such as vertical and horizontal array correlation length. The relationship between temporal fluctuations in the intensity of the surface reflected multi-path and array spatial coherence is investigated. To what extent receiver array statistics related to acoustic focusing follows environmental conditions such as wind speed and sea state are considered.

9:30

5aUW7. Sound propagation in a model ocean with internal waves: experiments and simulations. Likun Zhang, Santiago J. Benavides, and Harry L. Swinney (Dept. of Phys. and Ctr. for Nonlinear Dynam., Univ. of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199, lzhang@chaos.utexas.edu)

We conduct laboratory experiments and numerical simulations in a laboratory tank where sound waves and internal gravity waves are simultaneously generated and detected. The tank contains a density-stratified fluid with internal gravity waves produced by a wave generator. Acoustic sources and receivers are arranged such that the acoustic track crosses the internal wave beam. The internal wave energy density, energy flux, and total power are determined from particle imaging velocimetry measurements. The internal wave field is also computed in direct numerical simulations of the Navier-Stokes equations, using a parallelized finite-volume based solver code. The fluctuations in sound speed and intensity are determined as a function of the acoustic track location and the internal wave amplitude, frequency, phase, and modulation frequency. This research is designed to achieve a better understanding of sound propagation and scattering by internal waves in the oceans. [This research was supported by ONR MURI Grant N000141110701 (WHOI). L.Z. acknowledges the support of the 2013-14 ASA F. V. Hunt Postdoctoral Research Fellowship.]

9:45

5aUW8. Amplitude and phase fluctuations of the sound fled in vicinity of horizontally stratified perturbation in shallow water (parabolic equation modeling). Jixing Qin (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China) and Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru)

Based on the approach of vertical modes and the parabolic equation (PE) in horizontal plane, three-dimensional (3D) effects in propagation of sound signals are considered: structure variation of broadband signal in the presence of horizontal stratification, such as coastal wedge area, temperature front, and nonlinear internal waves. Main factor, influencing on sound propagation is dependence of horizontal refraction on frequency and mode number. The following effect are studied: interference between direct and reflected signals in multipath area of horizontal plane for different frequencies and mode numbers, different directions of amplitude and phase front, evolution of spectrum and shape of pulse. [Work was supported by RFBR and BSF.]

10:00–10:15 Break

10:15

5aUW9. An investigation into the bottom interface treatment in parabolic equation models utilizing split-step Fourier and finite-difference algorithms. Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Rm. 114, Monterey, CA 93943, kbsmith@nps.edu), David J. Thomson (733 Lomax Rd., Victoria, BC, Canada), and Paul Hursky (Heat, Light and Sound Res., Inc., San Diego, CA)

Recent analysis of parabolic equation model results computed using split-step Fourier algorithms have shown increasing phase errors at long range in simple, isospeed waveguides. Preliminary results suggest that these errors are due to the treatment of the density discontinuity at the water/bottom interface through the use of smoothing functions. In this work, several different approaches are investigated for improving the phase accuracy of the solution, including hybrid methods employing split-step Fourier and finite-difference algorithms, as well as equivalent bottom characterizations. The results are compared relative to their improvements in accuracy as well as computational efficiency.

10:30

5aUW10. Bottom interacting acoustics in the north Pacific. Ralph A. Stephen (Woods Hole Oceanogr. Inst., 360 Woods Hole Rd., Woods Hole, MA 02543-1592, rstephen@whoi.edu), Peter F. Worcester (Scripps Inst. of Oceanogr., La Jolla, CA), Ilya A. Udovydchenkov (Woods Hole Oceanographic Inst., Woods Hole, MA), and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

During the 2004 Long-range Ocean Acoustic Propagation Experiment (LOAPEX), a new class of acoustic arrivals was observed on ocean bottom seismometers (OBSs) for ranges from 500 to 3200 km. The arrivals were called deep seafloor arrivals (DSFAs), because they were the dominant arrivals on the ocean bottom seismometers (OBSs), but were very weak on the deep vertical line array (Deep VLA), located above 750 m from the seafloor. Stephen *et al*. [JASA **134**, 3307–3317 (2013)] attributed some of these arrivals to bottom-diffracted, surface-reflected (BDSR) energy that scattered from a seamount near the Deep VLA and subsequently reflected from the sea surface before arriving at the OBSs. In the Ocean Bottom Seismometer Augmentation in the North Pacific (OBSANP) Experiment in June to July 2013, we returned to the Deep VLA site with a near-seafloor Distributed Vertical Line Array (DVLA) that extended upward 1000 m from the seafloor and 12 OBSs. We transmitted to the instruments with a ship-suspended J15-3 acoustic source. The receiver locations and transmission program were designed to test the hypothesis that DSFAs correspond to BDSR energy, to further define the characteristics of the DSFAs, and to understand the conditions under which DSFAs are excited and propagate.

10:45

5aUW11. Weakly dispersive modal pulse propagation in the North Pacific Ocean. Ilya A. Udovydchenkov (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., MS #9, Woods Hole, MA 02543, ilya@whoi.edu), Michael G. Brown (Appl. Marine Phys., Rosenstiel School of Marine and Atmospheric Sci., Miami, FL), Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA), James A. Mercer, Rex K. Andrew (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Bruce M. Howe (Dept. of Ocean and Resources Eng., Univ. of Hawai, Honolulu, HI), and John A. Colosi (Dept. of Oceanogaphy, Naval Postgrad. School, Monterey, CA)

The propagation of weakly dispersive modal pulses is investigated using data collected during the 2004 long-range ocean acoustic propagation experiment (LOAPEX). Weakly dispersive modal pulses are characterized by weak dispersion- and scattering-induced pulse broadening; such modal pulses experience minimal propagation-induced distortion and are thus well suited to communications applications. In the LOAPEX environment, modes 1, 2, and 3 are approximately weakly dispersive. Using LOAPEX observations it is shown that, by extracting the energy carried by a weakly dispersive modal pulse, a transmitted communications signal can be recovered without performing channel equalization at ranges as long as 500 km; at that range a majority of mode 1 receptions have bit error rates (BERs) less than 10%, and 6.5% of mode 1 receptions have no errors. BERs are estimated for low order modes and compared with measurements of signal-to-noise ratio (SNR) and modal pulse spread. Generally, it is observed that larger modal pulse spread and lower SNR result in larger BERs. [Work supported by ONR.]

11:00

5aUW12. Weakly dispersive modal pulses in long-range underwater acoustic communications. Ilya A. Udovydchenkov (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., MS #9, Woods Hole, MA 02543, ilya@whoi.edu)

Weakly dispersive modal pulses are contributions to acoustic wave field from individual modes that are characterized by weak dispersion- and scattering-induced broadening. These pulses experience little propagation-induced distortions and are suitable for communications applications. This work investigates, using numerical simulations, how to design a communication system that takes advantage of the physics of weakly dispersive modal pulses. Two groups of weakly dispersive modal pulses are identified in typical mid-latitude ocean environments: the lowest order modes, and mode numbers with the waveguide invariant being near-zero (often around mode 20 at 75 Hz). This work analyzes the source and receiving array requirements for achieving low bit error rates (BERs) in a binary communication without performing channel equalization. It is shown that low BERs are achieved with only 3 hydrophones for mode 1 processing at 500 km and with 30 hydrophones for mode 20 at 400 km range with good signal-to-noise ratio (SNR). It is demonstrated that if depths of hydrophones are allowed to vary with the source-receiver distance, 2 hydrophones are often sufficient to achieve low BERs even with intermediate mode numbers. Thus, full modal resolution is often unnecessary to achieve low BERs. The effects of variable SNR are also studied. [Work supported by ONR.]

11:15

5aUW13. Leveraging spatial diversity to mitigate interference in underwater acoustic communication networks. James McGee (Code 15 Sensors and SONAR Systems, Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, james.a.mcgee@navy.mil), Peter Swaszek (Elec., Comput. and Biomedical Eng., Univ. of Rhode Island, Kingston, RI), and Josko Catipovic (Code 15 Sensors and SONAR Systems, Naval Undersea Warfare Ctr., Newport, RI)

Many acoustic communication channels suffer from interference which is neither narrowband nor impulsive. This relatively long duration partial band interference can be particularly detrimental to system performance. To address the problem of underwater communications over a time-varying

multipath channel in the presence of partial band interference, we examine a receiver which leverages the spatial diversity implicit in a network of geographically distributed hydrophones due to the slow speed of sound propagation. The network consists of multiple cabled hydrophones which receive communication signals from multiple users in addition to interfering signals from active sonars and marine mammals. The partial band interference corrupts different portions of the received signal depending on the relative position of the interferers, information source and receivers. The need for explicit time alignment and channel compensation due to the differing propagation paths between sources and receivers complicates combining the received signals. After surveying recent work in interference mitigation and orthogonal division multiplexing as background motivation, we examine the problem of combining information from such differentially corrupted signals through simulation.

11:30

5aUW14. Low frequency acoustic communication and the waveguide modal behavior during Shallow Water 2006 experiment. Mohsen Badiey and Aijun Song (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

Low frequency (i.e., <500 Hz) acoustic intensity fluctuations in the presence of internal waves has been studied in some detail during the past two decades. However, not much has been reported on the relationship between the modal behavior of the waveguide and the potential for long-range communications. Previously, we have shown significant performance degradation at higher frequencies (i.e., 800 and 1600 Hz) while an internal wave packet travels through a source-receiver track. Here, we further examine the acoustic transmissions at lower frequencies (i.e., 100 and 200 Hz) during the same internal wave event. Significant intensity fluctuations at these frequencies can be explained by modal analysis. We also report acoustic communication performance at multiple frequencies (i.e., 100, 200, 800, and 1600 Hz). The frequency dependency is analyzed with the focus on modal behaviors to explain the performance variation of acoustic communication during the passage of the internal waves.

11:45

5aUW15. Multiple input multiple output underwater communication based on differential amplitude phase shift keying modulation. Xu Xia and Jingwei Yin (College of Underwater Acoust. Eng., Harbin Eng. Univ., Rm. 801, Acoust. Bldg., No.145 Nantong St., Harbin, Heilongjiang 150001, China, heaven663@163.com)

The MIMO-OFDM combining time, spatial, and frequency diversity can effectively improve channel capacity and transmission efficiency of underwater acoustic (UWA) communication system. Space-time coding needs a large number of pilot signals to estimate UWA channel at the receiving end, which increases the complexity of the systems and limits the communication rate. For this, space-time coding combined with differential amplitude phase shift keying modulation (DAPSK) is proposed in this paper. It can complete the decoding without any prior knowledge of UWA channel, reducing the complexity of the system, saving channel resources and improving the transmission efficiency. Simulation analysis on UWA MIMO-OFDM systems shows this algorithm is feasible, which provides a feasible method for high-speed transmission in UWA communication.