Session 1aAA

Architectural Acoustics, Noise and Signal Processing in Acoustics: Spherical Array Processing for Architectural Acoustics

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Chair's Introduction—8:00

Invited Papers

8:05

1aAA1. Some spatial analysis of the room impulse response. Jens Meyer (mh Acoustics, 38 Meade Rd., Fairfax, VT 05454, jmm@mhacoustics.com) and Gary W. Elko (mh Acoust., Summit, NJ)

The room impulse response (RIR) and measures derived from it have been a powerful tool for room characterization. Traditionally, these measures are based on time domain characteristics like reverberation time or clarity. By recoding a group of RIRs with microphone arrays, one can add the spatial dimension to the RIR. Spherical microphone arrays are of special interest since they can give equal weight to all directions. In this paper, we explore an adaptive cardioid beampattern algorithm for its suitability as a tool for the spatial analysis of RIRs. The algorithm uses the spherical harmonic base patterns ("Eigenbeams") of a spherical microphone array to form and steer a cardioid beampattern. The steering is automatically adjusted to minimize the output power in a specific time region. Due to the cardioid beampatterns sharp null in the beampattern, an adaptive cardioid algorithm can provide detailed spatial information on reflections as a function of time for impulse responses.

8:25

1aAA2. A new technique to measure directional impulse responses using a 64-channel spherical microphone. Jacob Adelgren and David T. Bradley (Phys. + Astronomy, Vassar College, 124 Raymond Ave., Poughkeepsie, NY 12604, jaadelgren@vassar.edu)

A spherical microphone is an array consisting of numerous microphones arranged almost uniformly on the surface of a rigid sphere. Recently, spherical microphone arrays have received attention in architectural acoustics for their ability to characterize the sound field over a full sphere, which can yield additional information about the spatial characteristics of the field, including field diffusivity and directionality of individual reflections. Typically, these arrays are used as real-time analyzers to capture sound pressure information. However, in architectural acoustics, an impulse response measurement is required to adequately characterize the space. An effective approach for utilizing spherical microphones, particularly those with a large number of channels, to capture directional impulse responses has yet to be realized. The current project has utilized a 64-channel spherical microphone to carry out the integrated impulse response measurement technique using a sine-sweep signal. MATLAB has been used to evaluate sound fields in a reverberant chamber using a variety of directionality patterns (e.g., omnidirectional, cardioid, and figure-of-eight). Additionally, work is in progress to allow for arbitrary beamforming and other directionality patterns. Results and analysis will be presented.

8:45

1aAA3. An open-source spherical microphone array design. John Granzow, Tim O'Brien, Darrell Ford, Yoo H. Yeh, Yoomi Hur (CCRMA, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305, granzow@ccrma.stanford.edu), Darius Mostowfi (Sigma Eng., San Carlos, CA), and Jonathan S. Abel (CCRMA, Stanford Univ., Stanford, CA)

We present an open-source design for producing a spherically baffled, 32-channel microphone array having elements aligned with the vertices and face centers of a dodecahedron, and fabricated using additive manufacturing (3-D printing). The design approach emphasized low cost, assembly ease, and acoustic integrity. Mechanical, electrical, and acoustical design considerations are discussed, as are assembly and calibration details. The baffle is a 10-cm-radius sphere, built from separable hemispheres, and supported by a 2-cm-diameter cylindrical stand that serves as a conduit for the microphone signal cables. Microphone capsules are mounted on preamp boards, which provide balanced line level outputs, and press fit to the baffle. The performance of the array is characterized, and an example application to spatial room impulse response measurement is provided. Design documents, including the enclosure model and preamp electrical design and board layout, are provided at https://ccrma.stanford.edu/~granzow/sphericalmicarray/

9:05

1aAA4. Equivalence of plane wave and spherical harmonics rendering of binaural room impulse response. Zamir Ben-Hur, Jonathan Sheaffer, Boaz Rafaely (Elec. & Comput. Eng., Ben-Gurion Univ. of the Negev, Be'er Sheva, Be'er Sheva 8410501, Israel, zami@post.bgu.ac.il)

Binaural technology has various applications in virtual acoustics, architectural acoustics, tele-communications, and auditory science. One key element in binaural technology is the binaural room impulse response (BRIR), which represents a continuum of plane waves spatially filtered by head related transfer functions (HRTFs). Such BRIRs can be rendered from spherical microphone array recordings and free-field HRTFs, either in the space domain using plane-wave composition or in the spherical-harmonics domain using order-limited spherical harmonics representation of the sound field. While these approaches have been individually employed in a number of recent studies, it appears that the literature does not offer a comprehensive analysis or a theoretical framework relating the two representations with respect to binaural reproduction and perception. In this paper, we provide a mathematical analysis showing that when certain sampling conditions are maintained, the plane-wave and spherical-harmonics representations are equivalent. Further, we show that under these conditions, resulting binaural signals are independent of the employed spatial sampling schemes. The analysis is complemented by a listening experiment, in which both plane-wave and spherical-harmonics representations are perceptually evaluated for different spatial sampling schemes and spherical harmonic orders.

9:25

1aAA5. Binaural perception of direct and reverberant sound fields rendered with mixed-order spherical harmonics. Jonathan Sheaffer and Boaz Rafaely (Elec. and Comput. Eng., Ben-Gurion Univ., Beer-Sheva, Beer-Sheva 8410501, Israel, sheaffer@ee.bgu.ac.il)

Binaural responses can be rendered from a plane-wave decomposition of a measured or a modeled sound field, spatially integrated with free-field head-related transfer functions. When represented in the spherical-harmonics domain, the decomposition order reflects the maximum spatial resolution, which is limited by the number of microphones in the spherical array. Recent studies suggest a direct relationship between decomposition order and perceptual attributes such as localization blur, timbre, and the sense of externalization. Insofar, studies have employed plane wave density functions in which the different components of the sound field were uniformly decomposed at a single spherical-harmonics order. This work is concerned with binaural signals in which the direct and reverberant parts of the sound field are decomposed at different spherical-harmonics orders. The direct component of the sound field carries significant directional information utilized by the auditory system. Therefore, changing the spherical-harmonics order of the direct component is expected to have a different perceptual impact compared to changing the spherical-harmonics order of the reverberant part. Listening experiments are employed to study the perception of such mixed-order representations in context of sound localization and auditory distance perception.

9:45–10:00 Break

10:00

1aAA6. Investigation of listener envelopment using spherical microphone array measurements and ambisonics playback. David A. Dick and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dad325@psu.edu)

An important aspect of overall room impression is listener envelopment (LEV), the sense of being immersed in a sound field. Current LEV objective metrics are primarily based on lateral reflections, typically measured with a figure-of-eight microphone. The purpose of this study was to investigate LEV using measured impulse responses (IRs) taken with an Eigenmike 32-element spherical microphone array as an initial step toward creating a new metric for LEV. The spherical array enables a spatial analysis with higher resolution than traditional methods. Impulse response measurements were made in the Peter Kiewit Concert Hall in Omaha, NE, in several seat positions and hall absorption settings. Auralizations were generated by convolving the measured IRs with anechoic music excerpts and then processing the signals for third-order ambisonics playback. The signals were played over an array consisting of 32 loudspeakers in an anechoic chamber. A subjective study was run in which musically trained listeners rated the LEV of the stimuli. Beamforming techniques were used to analyze the IRs to better understand the spatial and temporal factors that contribute to the perception of envelopment. Results will be presented, which correlate the objective measurements to the subjective ratings. [Work supported by NSF Grant 1302741.]

10:20

1aAA7. The effect of playback method on listeners' judgments of concert hall auralizations. Samuel W. Clapp (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Arcisstr. 21, München 80333, Germany, samuel.clapp@tum.de), Anne E. Guthrie (Arup Acoust., New York, NY), Jonas Braasch, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Previous studies of the perception of concert hall acoustics have generally employed one of two methods for soliciting listeners' judgments. The first is to have listeners rate the sound of a hall while physically present in that hall. The second is to record the acoustics of a hall and later simulate those acoustics in a laboratory setting. While the first method offers a completely authentic presentation of the concert experience, the second allows for more direct comparisons between different spaces and affords the researcher greater control over experimental variables. Higher-order spherical microphone arrays offer a way to capture the spatial components of a concert hall's acoustics, which can then be reproduced using a loudspeaker array. In this study, eight different concert and recital halls were measured using both a spherical microphone array and binaural dummy head as receivers. Listeners were then presented with auralizations of the halls using an ambisonic loudspeaker array and headphones, and asked to rate the halls based on subjective preference and on similarity to one another. The responses were analyzed using multidimensional scaling methods in order to examine the effect of the auralization system on the listeners' judgments.

10:40

1aAA8. Spatial analysis of sound fields in rooms using spherical MIMO systems. Hai Morgenstern (Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, POB 653, Beer Sheva 84105, Israel), Markus Noisternig (UMR IRCAM-CNRS-UPMC, Paris, France), and Boaz Rafaely (Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, Beer Sheva, Israel, br@bgu.ac.il)

The perception of sound by human listeners in a room has been shown to be affected by the spatial attributes of the sound field. These spatial attributes have been studied using microphone and loudspeaker arrays separately. Systems that combine both loudspeaker and microphone arrays, termed multiple-input multiple-output (MIMO) systems, facilitate enhanced spatial analysis compared to systems with a single array, thanks to the simultaneous use of the arrays and the additional spatial diversity. Using MIMO systems, room impulse responses (RIRs) can be presented using matrix notation, which enables a unique study of a sound field's spatial attributes, employing methods from linear algebra. For example, a matrix's rank and null space can be studied to reveal spatial information on a room, such as the number of dominant room reflections and their direction of arrival to the microphone array and the direction of radiation from the loudspeaker array. In this contribution, a theory of the spatial analysis of a sound field using a MIMO system comprised of spherical arrays is developed and a simulation study is presented. In the study, tools proposed for processing MIMO RIRs with the aim of revealing valuable information about acoustic reflections paths are evaluated.

11:00

1aAA9. Practical approach to forward and reciprocal sequential array measurements. Johannes Klein and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, johannes.klein@akustik.rwth-aachen.de)

Room impulse response measurements including directivity are becoming a vital component of room acoustics. The results of these measurements allow for the room acoustical analysis regarding specific source receiver combinations and enable the realistic simulation of acoustical scenarios in virtual environments. Besides the widely applied microphone-arrays, electroacoustic sources with a steerable directivity are necessary to retain all degrees of freedom during the measurements. Due to the relatively large dimensions of the transducers, the number of possible physical transducers in a loudspeaker-array is very limited, resulting in a low spatial resolution. A solution to this problem are sequential measurement methods, which virtually enlarge the number of transducers by moving the source during the measurement and superposing the results. The attainable high spatial resolution is ideal for substituting the most complex source or receiver directivity in forward or reciprocal measurements. However, sequential measurement methods are very susceptible to time variances, which diminish the achievable accuracy. This study is an experimental approach with various source receiver combinations to evaluate the practically achievable accuracy in forward and reciprocal sequential loudspeaker-array measurements in different environments and the corresponding sources of error.

11:20

1aAA10. Spatial aliasing study using a high-resolution radiation measurement of a violin. Noam R. Shabtai, Gottfried Behler, and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, nsh@akustik.rwth-aachen.de)

Sound source radiation patterns can be employed in a virtual acoustic system in order to improve the realistic experience of the listener. Many studies were described in the literature, in which spherical microphone arrays and spherical harmonics were used to capture the radiation pattern of musical instruments. In these studies, however, a limited number of microphones is used due to technical reasons that involve a non-repeatable excitation by the human player and the corresponding requirement to capture the radiation in one single excitation. Up to now, the radiation pattern is measured without a reference that describes the degree of spatial aliasing caused by the limited number of microphones. This work presents a high-resolution spatial sampling of the radiation pattern of an electrically excited violin. An analytical measure to the degree of spatial aliasing is represented and calculated for each number of microphones, using the high-resolution measurement as a reference.

11:40

1aAA11. Array processing methods for the determination of reflection properties of architectural surfaces. Markus Müller-Trapet and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, mmt@akustik.rwth-aachen.de)

It has become popular in architectural acoustics to use microphone arrays, very often in spherical arrangements, to capture and analyze the sound field in rooms. This contribution will present a hemispherical receiver array, designed to obtain information about sound reflection from architectural surfaces, preferably *in-situ*. Similarly to previous studies, the analysis in the spherical harmonics (SH) domain is favored, with the additional challenge of data available on a hemisphere instead of a complete sphere. This problem is solved by obtaining orthonormal base functions on the hemisphere. As application cases, spherical beamforming for the determination of reflection factors will be presented as well as scattering near-field holography in order to determine the diffusion coefficient of small samples. Results from numerical and experimental case studies will be discussed.

Session 1aAB

Animal Bioacoustics: Bioacoustics and Behavior

Samuel L. Denes, Chair
Acoustics, Pennsylvania State Univ., 116 Applied Science Building, University Park, PA 16802

Contributed Papers

8:30

1aAB1. Individually distinctive parameters in the upcall of the North Atlantic right whale (*Eubalaena glacialis*). Jessica A. McCordic and Susan E. Parks (Biology, Syracuse Univ., 114 Life Sci. Complex, Syracuse, NY 13244, jamccord@syr.edu)

According to the source-filter hypothesis proposed for human speech, physical attributes of the vocal production mechanism combine independently to result in individually distinctive vocalizations. In the case of stereotyped calls with all individuals producing a similar frequency contour, the filtering of the signal resulting from the shape and size of the vocal tract may be more likely to contain individually distinctive information than parameters measured from the fundamental frequency resulting from the vibrating source. However, the formant structure resulting from such filtering has been historically undervalued in the majority of studies addressing individual distinctiveness in non-human species. The upcall of the North Atlantic right whale (*Eubalaena glacialis*) is characterized as a stereotyped contact call, and visual inspection of upcall spectrograms confirms presence of a robust formant structure. Here we present preliminary results testing individual distinctiveness of upcalls recorded from archival, suction cup mounted tags (Dtags). Parameters measured from the fundamental frequency contours as well as the formant structure of the calls are used in assigning upcalls to individual whales. These results provide a baseline for further development of acoustic detection techniques that could be used to noninvasively track movements of individual whales across habitats.

8:45

1aAB2. Exploring the vocal ontogeny of North Atlantic right whales. Holly Root-Gutteridge, Dana Cusano (Dept. of Biology, Syracuse Univ., Syracuse, NY 13244, hrootgut@syr.edu), Lisa Conger, Sofie Van Parijs (Northeast Fisheries Sci. Ctr., NOAA Fisheries, Woods Hole, MA), and Susan Parks (Dept. of Biology, Syracuse Univ., Syracuse, NY)

The vocal ontogeny of species can provide insight into their physical and social development and capacity for social learning. North Atlantic right whales (*Eubalaena glacialis*) are an endangered species with a complex range of vocalizations used in a wide range of social contexts. In this study, we systematically characterize developmental changes in sound production from birth through adulthood. Calls have been recorded as part of ongoing North Atlantic right whale research projects spanning 2001 through 2015, and include data from calves as young as 1 month of age. Data included recordings from single hydrophones, hydrophone arrays, and non-invasive digital acoustic recording tags. Only calls which could be confidently attributed to a specific whale of known age were used in the analysis. Developmental periods consisted of 0–3 months, 3–6 months, 6–12 months, and then discrete ages by year. Calls of both male and female calves were included in the analysis. Calls were also classified to age categories using discriminant function analysis (DFA) to determine whether a calf could be aged by its call. Analyses indicate a gradual maturation of sound production with increasing age of individuals.

9:00

1aAB3. Objective analysis of vocal sequences using nested self-organizing maps. Eduardo Mercado (Dept. of Psych., Univ. at Buffalo, Buffalo, NY 14260, emiii@buffalo.edu)

Traditional approaches to analyzing vocal sequences typically involve identifying individual sound units, labeling identified sounds, and describing the regularity of label sequences [A. Kershenbaum *et al*., "Acoustic sequences in non-human animals: A tutorial review and prospectus," Biol. Rev. (2014)]. Although this method can provide useful information about the structure of sound sequences, the criteria for determining when distinct units have been successfully classified are often subjective and the temporal dynamics of sound generation are usually ignored. Self-organizing maps (SOMs) provide an alternative approach to classifying inputs that does not require subjective sorting or isolation of units. For instance, SOMs can be used to classify fixed duration frames sampled from recordings. Once an SOM has been trained to sort frames, the temporal structure of a vocal sequence can be analyzed by training a second SOM to sort spatiotemporal patterns of activation within the frame-sorting SOM. Analyzing humpback whale "song" using this technique revealed that: (1) perceptually warped spectra from frames varied uniformly along several continua; (2) a subset of frame patterns (sound types) was more prevalent; and (3) produced features varied systematically as a function of sequential position within a song for some sounds, but not others.

9:15

1aAB4. Comparison of managed care and wild walrus source characteristics. Samuel L. Denes (Biology, Syracuse Univ., 116 Appl. Sci. Bldg., University Park, Pennsylvania 16802, sld980@psu.edu), Jennifer L. Miksis-Olds (Appl. Res. Lab., The Penn State Univ., University Park, PA), Dave Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR), Eric Otjen (Animal Care, SeaWorld San Diego, San Diego, CA), and Ann E. Bowles (Hubbs-SeaWorld Res. Inst., San Diego, CA)

Male Pacific walruses perform acoustic displays while in rut. The purpose of these displays is unknown but are hypothesized to be for territory defense or mate advertisement. Understanding source characteristics will allow the estimation of perceptibility by conspecifics. The displays occur in the Bering Sea in late winter where direct human observation is difficult. Working with an animal in managed care provided the ability to make direct observations of a male producing breeding vocalizations and the direct calculation of source level. Source characteristics from recordings of managed care and wild walruses were analyzed. The mean peak source level of the impulsive knocks produced by the managed care male was 183 dB (re: 1 μPa). The mean peak source level from the wild recordings was 177 dB (re: 1 μPa). For both wild and managed care vocalizations, a significant relationship between ambient noise level and source level was identified. An increase of approximately 5 dB in source level was found for an increase in 10 dB of noise level.

9:30

1aAB5. Bleats are universally used by ungulates for mother-offspring communication but what use do they serve for toads and pandas? David Browning (139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com)

Bleats are usually associated with sheep and goats, but are universally used by ungulates for mother-offspring communication. These short, relatively simple vocalizations are used, depending on circumstance, for localization, identification, emotion, and guidance. Interestingly, there are at least two other cases of bleat users. The Australian bleating toad appears to use them as an efficient means of continuous vocalization, while pandas use bleats as part of a repertoire of jungle talk to communicate in dense bamboo thickets.

9:45

1aAB6. Mobility of dinoflagellates measured by high-frequency ultrasound. Hansoo Kim (Ocean System Eng., Jeju National Univ., Ocean Science College 4, Jeju 690-756, South Korea, hansoo5714@naver.com), Tae-Hoon Bok (Phys., Ryerson Univ., Toronto, ON, Canada), Kweon-Ho Nam, Juho Kim, Dong-Guk Paeng (Ocean System Eng., Jeju National Univ., Jeju, South Korea), So-Jeong An, and Joon-Baek Lee (Earth and Marine Sci., Jeju National Univ., Jeju, Korea (the Republic of))

The importance of phytoplankton contributing more than 50% of the global amount of photosynthesis has been emphasized for a long time. Sometimes, the over-growth of phytoplankton causes the negative influence such as red-tide phenomenon on marine ecological environments. Therefore, the measurement of the mobility of phytoplankton is important. In this study, the mobility of the benthic dinoflagellate, *Amphidinium carterae* Hulburt (*A. Carterae*) incubated by f/2 medium was investigated using high-frequency ultrasound. Backscattering signal from *A. Carterae* was measured for 2 s in every 2 min by a 40-MHz ultrasound transducer, and the integrated backscattering power calculation was followed. The mobility of *A. carterae* in response to the light was illustrated by M-mode image of the echoed signals. The mobility of *A. carterae* was estimated to about 0.4 mm/s for the upward movement in response to light, while its sedimentation rate was measured to about 0.1 mm/s in a dark environment. This study suggests that mobility of benthic dinoflagellates responding to light can be measured by M-mode imaging of high-frequency ultrasound. (This research was a part of the project titled "Measurement of cells division and photosynthesis of phytoplankton using ultrasound", funded by the Ministry of Oceans and Fisheries, Korean.)

MONDAY MORNING, 18 MAY 2015 BALLROOM 1, 10:00 A.M. TO 1:00 P.M.

Session 1aED

Education in Acoustics: Hands-On Acoustics Demonstrations for Middle- and High-School Students

Cameron T. Vongsawad, Cochair
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Kent L. Gee, Cochair
Brigham Young University, N243 ESC, Provo, UT 84602

Invited Paper

10:00

1aED1. Auditory illusion demonstrations as related to prehistoric cave art and Stonehenge. Steven J. Waller (Rock Art Acoust., 5415 Lake Murray Blvd. #8, La Mesa, CA 91942, wallersj@yahoo.com)

Auditory illusions will be demonstrated, and their relevance to prehistoric cave art and Stonehenge revealed. In the ancient past, when the wave characteristics of sound were not understood, virtual sound effects arising from complex sound wave interactions (echoes, reverberations, interference patterns, etc.) were misinterpreted as invisible beings (echo spirits, thunder gods, sound absorbing bodies, etc.) as described in ancient myths around the world. In this session, live hands-on demonstrations will be given to small groups of students who can experience for themselves these types of auditory illusions. Participants will get to experience various sounds and will be given the task of interpreting what the sounds are, first blindfolded, then with visual cues. (Previous student reactions have included "Whoa!," "Wow!," "Amazing!," "Cool!," and "What??.") These scientifically conducted experiments show how various ambiguous sounds can be interpreted in more than one way—like optical illusions—and thus can help in understanding our ancestors' reactions to sounds they considered mysterious and spooky. These discoveries are just a few examples of research findings that are springing from the new field of Archaeoacoustics. See https://sites.google.com/site/rockartacoustics/ for further examples.

Session 1aPA

Physical Acoustics: General Topics in Physical Acoustics I

Brian E. Anderson, Chair
Geophysics Group (EES-17), Los Alamos National Laboratory, MS D446, Los Alamos, NM 87545

Contributed Papers

8:30

1aPA1. The vibroacoustical environment in two nuclear reactors. Joshua Hrisko, Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Graduate Program in Acoust., Appl. Res. Lab, State College, PA 16804, jqh5657@psu.edu), Robert W. Smith (Appl. Res. Lab., Penn State, State College, PA), James A. Smith, and Vivek Agarwal (Fundamental Fuel Properties, Idaho National Lab., Idaho Falls, ID)

Laboratory experiments have suggested that thermoacoustic engines can be incorporated within nuclear fuel rods. Such engines would radiate sounds that could be used to measure and acoustically-telemeter information about the operation of the nuclear reactor (e.g., coolant temperature or fluxes of neutrons or other energetic particles) or the physical condition of the nuclear fuel itself (e.g., changes in porosity due to cracking, swelling, evolved gases, and temperature) that are encoded as the frequency and/or amplitude of the radiated sound [IEEE Measurement and Instrumentation **16**(3), 18–25 (2013)]. For such acoustic information to be detectable, it is important to characterize the vibroacoustical environments within reactors. We will present measurements of the background noise spectra (with and without coolant pumps) and reverberation times within the 70,000 gallon pool that cools and shields the fuel in the 1 MW research reactor on Penn State's campus using two hydrophones, a piezoelectric projector, and an accelerometer. Background vibrational measurement taken at the 250 MW Advanced Test Reactor, located at the Idaho National Laboratory, from accelerometers mounted outside the reactor's pressure vessel and on plumbing, will also be presented to determine optimal thermoacoustic frequencies and predict signal-to-noise ratios under operating conditions. [Work supported by the U.S. Department of Energy.]

8:45

1aPA2. Using helium as the working fluid to improve efficiency of high-frequency thermoacoustic engines. Nathaniel Wells (Phys., Utah Valley Univ., Orem, TX) and Bonnie Andersen (Phys., Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84057, bonniem@uvu.edu)

Previous work on thermoacoustic engines with bottle-shaped resonators has been done to improve performance by varying geometric parameters, using air as the working fluid. This study is focused on transitioning from air to helium for the working fluid to further improve device performance. The theoretical ratio of efficiencies was derived for the two operating fluids. The existing engine was redesigned for evacuating the air and introducing helium into the resonator and six different types of heat shrink tubing used to hold the heat exchangers in place were tested for effectiveness with a vacuum and ease of removal. The optimal stack masses for this engine operating with air and helium were theoretically estimated and tested with air using six different stack masses from 50 to 62 mg. The resonator had a cavity with a length of 10 cm and ID of 4.13 cm and a neck with a length of 2.62 cm and ID of 1.91 cm and used steel wool for the stack material. The engine was supplied 12 W from a heating element applied above the hot heat exchanger in the neck. The engine was allowed to run for 40 s after the temperature had reached a steady state and the acoustic pressure at the bottom of the cavity was measured. The optimal amount of stack in air was found to be 56 mg, and the acoustic pressure was 206 Pa, Pk-Pk. The adhesive heat shrink tubing was found to be the most effective for use with helium and ease of removal.

9:00

1aPA3. Differences in atmospheric absorption coefficients between ANSI/ASA S1.26-2014 and an updated model. Erik A. Petersen and Victor W. Sparrow (Graduate Program in Acoust., Penn State Univ., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16801, eap206@psu.edu)

Absorption coefficient predictions from the ANSI/ASA standard S1.26-2014 and Sutherland and Bass 2004 [Sutherland, *et al*., J. Acoust. Soc. Am. **115**(3), 2004] are compared for 125, 250, 500, and 1000 Hz pure tones ranging in elevation from 0 to 10 km. The differences in absorption mechanisms as well as the assumed atmospheric profiles are considered. Calculated using their respective profiles, the two models differ by 1–4 dB/km. Additionally, cumulative absorption over a 10 km vertical path is calculated under several conditions; ANSI with ANSI profile, ANSI with S&B profile, and S&B with S&B profile. Comparing the second and third case shows a difference of 0.5–3 dB, which characterizes model dependent differences by evaluating both models using the same profile. A comparison of the first and second cases yields a difference of 0.5 to 10 dB, indicating a strong dependence on the assumed atmospheric profile. To achieve consistent predictions over a 10 km path the layer thickness should be no greater than 1 km. [The opinions, findings, conclusions, and recommendations expressed here are those of the authors and do not necessarily reflect the views of sponsors of the ASCENT Center of Excellence including the Federal Aviation Administration.]

9:15

1aPA4. High frequency oblique-angle acoustic reflections from an air-snow interface. Donald G. Albert, Arnold J. Song, and Zoe R. Courville (ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755, donald.g.albert@usace.army.mil)

Fresh natural snow is a difficult material to characterize, as any mechanical interaction is likely to damage the fragile pores and grain bonds. Because acoustic waves are sensitive to the porous material properties, they potentially can be used to measure snow properties in a non-destructive manner. Such methods have already been demonstrated on cohesive porous materials including manufactured foams, porous metals, and sintered glass beads. As a first step toward developing a portable method that can be used outdoors, we conducted high frequency oblique-angle reflection measurements on snow samples in a cold room. We compare the acoustically derived parameters with microcomputerized tomography (CT) methods and with standard (but destructive) laboratory measurements. [This research funded by the U.S. Army Corps of Engineers.]

9:30

1aPA5. Comparisons between physics-based, engineering, and statistical learning models for outdoor sound propagation. Nathan J. Reznicek, Carl R. Hart, D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, carl.r.hart@usace.army.mil), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), and Edward T. Nykaza (U.S. Army Engineer Res. and Development Ctr., Champaign, IL)

Many outdoor sound propagation models exist, ranging from highly complex physics-based simulations to simplified engineering calculations. More recently, highly flexible statistical methods have become available, which offer the capability to predict outdoor sound propagation, given a training dataset. Among the variety of modeling approaches, the range of incorporated physics varies from none, as in the statistical learning methods, to comprehensive considerations for physical models. Engineering methods vary in the level of incorporated physics, oftentimes resorting to heuristic or approximate approaches. In order to compare the capability of engineering and statistical learning models, one particular physics-based model is used for outdoor sound propagation predictions, namely, a Crank-Nicholson parabolic equation (CNPE) model. Narrowband transmission loss values predicted with the CNPE, based upon a simulated dataset of meteorological, boundary, and source conditions, act as simulated observations. Among the engineering models used in the comparisons are the Harmonoise propagation model and the ISO 9613-2 method. Among the statistical learning methods used in the comparisons is a random forest regression model. Metrics such as the root-mean-square error and the skill score are computed for both the engineering models and statistical learning models.

9:45

1aPA6. Use of evanescent plane waves for low-frequency energy transmission across material interfaces. Daniel C. Woods, J. S. Bolton, and Jeffrey F. Rhoads (School of Mech. Eng., Purdue Univ., 585 Purdue Mall, School of Mech. Eng., West Lafayette, IN 47907, woods41@purdue.edu)

The transmission of sound across high-impedance difference interfaces, such as an air-water interface, is of significant interest for a number of applications. Sonic booms, for instance, may affect marine life, if incident on the ocean surface, or impact the integrity of existing structures, if incident on the ground surface. Reflection and refraction at the material interface, and the critical angle criteria, generally limit energy transmission into higher-impedance materials. However, in contrast with classical propagating waves, spatially decaying incident waves may transmit energy beyond the critical angle. The inclusion of a decaying component in the incident trace wavenumber yields a nonzero propagating component of the transmitted surface normal wavenumber, so energy propagates below the interface for all oblique incident angles. With the goal of investigating energy transmission using incident evanescent waves, a model for transmission across fluid-fluid and fluid-solid interfaces has been developed. Numerical results are shown for the air-water interface and for common air-solid interfaces. The effects of the incident wave parameters and interface material properties are also considered. For the air-solid interfaces, conditions can be found such that no reflected wave is generated, due to impedance matching among the incident and transmitted waves, which yields significant transmission increases over classical incident waves.

10:00–10:15 Break

10:15

1aPA7. Crank-Nicholson solution of the wide-angle parabolic equation for inhomogeneous moving media. D. K. Wilson and Vladimir E. Ostashev (U.S. Army Cold Regions Res. and Eng. Lab., Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil)

Solutions of the narrow-angle parabolic equation (PE) are essentially the same in non-moving and moving media, due to the validity of the effective sound-speed approximation. However, the wide-angle PE (PE) and its numerical solution are considerably more complicated for a moving than for a non-moving medium. Starting with a rigorously derived, wide-angle PE for propagation in an inhomogeneous moving medium, the Crank-Nicholson solution of this equation is shown to involve a pentadiagonal matrix, which is relatively inefficient to calculate. However, a high-frequency approximation of this equation, valid when the wavelength is small compared to the length scale of the inhomogeneities, leads to a particularly simple wide-angle PE, the solution of which involves a tridiagonal matrix and thus can be solved with only slight extensions to existing narrow-angle PE codes. Solutions of the wide-angle PE are illustrated with examples for sound propagation in the atmosphere. Comparisons to the narrow-angle PE exhibit close agreement at low propagation angles. At higher propagation angles, the refraction effects are found to be relatively unimportant.

10:30

1aPA8. Outdoor measurements of shock-wave propagation from exploding balloons. Sarah M. Young (Dept. of Phys., Brigham Young University-Idaho, Romney 118, Rexburg, ID 83460, sarahmyoung24@gmail.com), Kent L. Gee, Traclanne B. Neilsen, and Kevin M. Leete (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Previous anechoic measurements of exploding latex balloons filled with stoichiometric mixes of acetylene and oxygen revealed how these sources could be used to study weak-shock decay over relatively short ranges [M. B. Muhlestein *et al.*, J. Acoust. Soc. Am. 131, 2422–2430 (2012)]. This paper describes an experiment conducted using a larger balloon over much longer propagation ranges at the Bonneville Salt Flats, which represents a hard, flat, relatively homogeneous ground surface. Measurements of a 0.56 m balloon were made along a propagation radial from 0.31 m from the balloon surface to 1600 m. Data were collected at a sampling rate of 204.8 kHz using piezoresistive pressure gauges and Type-1 condenser microphones. Described are waveform and spectral characteristics, as well as comparisons of the peak pressure decay with the weak-shock model employed previously. Waveform inspection and the comparison indicate that weak shocks are present out to at least 305 m and the amplitude decay rate can be predicted reasonably well using the model. Deviations from the model may be evidence of Mach-like reflections [K. M. Leete *et al.*, Four Corners Ann. Meet. Am. Phys. Soc. 59, I7.00007 (2014)]. This work extends the previous laboratory experiments and serves as a foundation for further studies using this relatively low-cost source.

10:45

1aPA9. Mach reflections in the propagation of outdoor acoustic shocks generated by exploding balloons. Kevin M. Leete, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, KML@byu.edu), Sarah M. Young (Dept. of Phys. and Astronomy, Brigham Young Univ., Rexburg, Idaho), Tadd T. Truscott, and Jonathon R. Pendlebury (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT)

When a shock wave reflects off a rigid surface, for certain combinations of shock strength and incident angle to the surface a Mach reflection can occur. This is when the incident and reflected shock waves merge to create a stronger shock wave (called a Mach stem) that travels parallel to the reflecting surface and whose height grows with distance. This phenomenon has been studied extensively for large explosions and for steady shock waves, but is less understood for acoustic weak shocks, where current models for Mach stem formation and growth do not agree with experimental observations. A weak-shock propagation experiment has been conducted at the Bonneville Salt Flats using blast waves generated by acetylene-oxygen filled balloons located at a fixed height above the ground. Analysis of acoustic data at various distances from the source and high-speed camera footage both identify a merging of the direct and ground-reflected waves and the formation of a Mach stem at locations closer to the source than theory would otherwise predict.

11:00

1aPA10. Quantitative analysis of a frequency-domain nonlinearity indicator. Kyle G. Miller (Brigham Young Univ., 323 East 1910 South, Orem, UT 84058, millerthepillar@gmail.com), Brent O. Reichman, Kent L. Gee, and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

An ensemble-averaged, frequency-domain version of the Burgers equation can be rearranged in order to directly compare the effects of nonlinearity on the sound pressure level, with the effects of atmospheric absorption and geometric spreading on a decibel scale. This nonlinear effect is calculated using the quadspectrum of the pressure and the pressure squared waveforms.

A number of nonlinearity indicators have been developed based on this quadspectral term, but their use has been largely qualitative. Further analysis of the expression has resulted in a quantitative understanding of how a quadspectrum normalization, referred to as Q/S, can be tied to the frequency-dependent sound pressure level changes due to nonlinearity. This understanding is developed by analyzing how Q/S evolves for analytical scenarios—the Blackstock Bridging Function and the Mendousse Solution. The decibel change per shock formation distance calculated through Q/S accurately captures the growth and decay of the higher harmonics, indicating that the most significant changes in the normalized quadspectrum occur before sawtooth formation.

11:15

1aPA11. Underwater laser acoustic source control using shaped plasmas. Theodore G. Jones, Michael Helle, Dmitri Kaganovich, Antonio Ting (Plasma Phys., U.S. Naval Res. Lab., NRL Code 6795, 4555 Overlook Ave. SW, Washington, DC 20375, ted.jones@nrl.navy.mil), Michael Nicholas, David Calvo (Acoust., U.S. Naval Res. Lab., Washington, DC), Gregory DiComo, and James Caron (Res. Support Instruments, Inc., Lanham, MD)

NRL is developing an intense laser acoustic source using underwater shaped plasmas. Recent experiments include near-field acoustic source characterization using high energy lens-focused pulses of a Q-switched Nd:YAG 532 nm laser. The laser-generated plasma evolves into a piston expanding at supersonic speed, which launches an intense shock in the near field. The size and shape of this super-heated piston determines the acoustic waveform and energy spectral density (ESD). We have demonstrated the ability to change the ESD centroid from 15 kHz to a few MHz, with lower frequencies generated using highly elongated plasmas generated by a single laser pulse. We will discuss ongoing laser acoustic source experiments and research plans at NRL involving shaped underwater plasmas, including both demonstrated single-laser-pulse techniques and proposed two-laser-pulse techniques (T. G. Jones, *et al.*, "Two laser generation of extended underwater plasma," U.S. patent application 13/711,752). Two-laser-pulse acoustic generation hold promise for creating meter-scale plasmas, thereby lowering the acoustic ESD to the few-kHz frequency range, which is useful for long-range applications including sonar and communications. Acoustic source characterization includes acoustic waveform and directivity measurements using hydrophones sensitive from 1 Hz to 15 MHz. [This work was supported by NRL Base Funds.]

11:30

1aPA12. Chemical kinetics theory of pyrotechnic whistles. Gregory W. Lyons and Richard Raspet (National Ctr. for Physical Acoust., The Univ. of MS, NCPA, P.O. Box 1848, University, MS 38677-1848, gwlyons@go.olemiss.edu)

Pyrotechnic whistles are sound effect devices commonly used in fireworks and consist of a particular fuel-oxidizer mixture pressed into the bottom of a tube. When ignited, a loud sound is emitted with harmonic frequencies corresponding to standing-wave modes of the tube. A theory of pyrotechnic whistles is developed for a chemical kinetics feedback mechanism in a model combustion reaction. Normal modes are obtained for a cylinder with steady, uniform axial flow. Boundary conditions are derived for a state-dependent reaction rate in an infinitesimal combustion surface. Solutions are presented for the normal modes with respect to reaction rate and tube outlet impedance.

MONDAY MORNING, 18 MAY 2015 — KINGS 3, 8:00 A.M. TO 12:00 NOON

Session 1aPPa

Psychological and Physiological Acoustics and Animal Bioacoustics: Kinematic Hearing: Auditory Perception of Moving Sounds by Moving Listeners

W O. Brimijoin, Cochair
Institute of Hearing Research, Medical Research Council, 10-16 Alexandra Parade, Glasgow G31 2ER, United Kingdom

Michael Akeroyd, Cochair
MRC/CSO Institute of Hearing Research - Scottish Section, New Lister Building, Glasgow Royal Infirmary, Glasgow G31 2ER, United Kingdom

Cynthia F. Moss, Cochair
Psychological and Brain Sci., Johns Hopkins Univ., 3400 N. Charles St., Ames Hall 200B, Baltimore, MD 21218

Chair's Introduction—8:00

Invited Papers

8:05

1aPPa1. Rotating sound sources and listeners: Sound source localization is a multisensory/cognitive process. William Yost, Xuan Zhong, and Anbar Najam (Speech and Hearing Sci., ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

When sound sources or listeners rotate, the acoustic cues used for sound source localization change, but in the everyday world listeners perceive sound rotation only when the sound source rotates not when the listener rotates. That is, in the everyday world, sound source locations are referenced to positions in the environment (a world-centric reference system). The acoustic cues for sound source location indicate a sound source's location relative to the head (a head-centric reference system), not locations relative to the world. To compute world-centric locations of sound sources, the auditory spatial system must have information about the acoustic cues used for sound source location and cues about the position of the head. The use of visual and vestibular information in determining head position in sound rotation perception was investigated in four experiments. The experiments clearly show, for the first time, that sound source localization when sound sources and listeners rotate is based on acoustic and visual, and sometimes vestibular information. Sound source localization is not based just on acoustics. It is a multisensory process. [Research supported by an AFOSR grant.]

8:25

1aPPa2. Updating and orientation in auditory space. Daria Genzel (Div. of Neurobiology, Dept. Biology II, Univ. of Munich, Grosshaderner Str. 2, Munich 82152, Germany), Paul MacNeilage (German Ctr. for Vertigo and Balance Disord., University Hospital of Munich, Munich, Germany), and Lutz Wiegrebe (Div. of Neurobiology, Dept. Biology II, Univ. of Munich, Munich, Germany, lutzw@lmu.de)

For stable sound localization in space, a listener has to account for self-motion. This requires a head-to-world coordinate transformation by combining proprioceptive and vestibular inputs with the binaural cues for sound localization. Proprioceptive and vestibular information influence the evaluation of spatial auditory cues as indicated by the numerous proprioceptive and vestibular neural inputs into the auditory brainstem. First, we evaluate the relative influence of vestibular and proprioceptive cues on updating the world-centered position of auditory targets during active and passive head and/or body rotations in azimuth. Our results show that both vestibular and proprioceptive signals are used to update the spatial representations of auditory targets, but that vestibular inputs contribute more than proprioceptive inputs. Second, we explore the interplay of spatial audition and self-motion in subjects for which auditory space evaluation is of utmost importance, namely, for blind humans relying on echolocation. Again, the techniques and paradigm allow disentangling vestibular and proprioceptive components for effective orientation based on the auditory analysis of the echoes of self-generated sounds. Our data show how good human biosonar can get for spatial orientation and how self motion helps suppressing orientation biases by lateral walls and front-back confusions.

8:45

1aPPa3. Hearing motion in motion. Simon Carlile, Johahn Leung, Shannon Locke, and Martin Burgess (School of Medical Sci., Univ. of Sydney, F13, Anderson Stuart Bldg., Camperdown, Sydney, New South Wales 2006, Australia, simonc@physiol.usyd.edu.au)

The acoustic cues to auditory space are referenced to the head which moves through the world, itself composed of moving sources so that our sensation convolves source with self- motion. With the head still, velocity discrimination, a perceptual process, is related to static acuity via the minimum audible movement angle (MAMA). Yet, while we can accurately localize static sounds, we are much less sensitive to velocity, resorting to distance and time cues where available. Interestingly, when velocity changes as a step function, discrimination thresholds and the amount of post-transition stimulus required for detection is greater than the corresponding MAMA. This suggests that when the head is stationary, the window of temporal integration may vary according to the sound's velocity characteristics. We also have evidence that auditory representational momentum scales with velocity, not duration or distance. Facing and following a moving auditory source is an ecologically important behavior. Tracking exhibits on-line velocity correction for slow to moderate velocities ($< 80\,^\circ$/s) but at higher velocities reflects a more predictive mechanism. Patients with schizophrenia are impaired in their ability to track a moving auditory target, when compared with controls, despite having normal velocity perception when the head is not moving. The presence of other efference copy dysfunctions in schizophrenia suggests a key role for motor efference copy in the disambiguation of self and target motion.

9:05

1aPPa4. Smooth pursuit eye and gaze movements to moving auditory targets: Evidence for velocity-specific processing in the auditory system. Christina Cloninger, Justin T. Fleming, Paul D. Allen, William E. O'Neill, and Gary D. Paige (Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, christina_cloninger@urmc.rochester.edu)

Auditory motion perception remains poorly understood, in contrast with its well-established visual counterpart. Visual smooth pursuit (SP), a velocity-specific behavior, has been well-quantified during both head-fixed (*ocular* SP) and head-free (eye + head, or *gaze* SP) conditions. In contrast, auditory SP tracking has received little attention, despite its potential for demonstrating a motion-specific process in audition. We presented constant-velocity (10–40°/s), free-field auditory (0.2–20 kHz white noise), and visual (LED) targets to head-fixed or head-free subjects while recording *ocular* and *gaze* SP responses, respectively. To control for possible SP in the absence of a target, subjects were asked to recreate auditory trajectories after priming with a set of auditory ramps (practiced SP). We found that *ocular* auditory SP is consistently higher in gain than practiced SP, but variable and lower than visual SP. Further, the gain of auditory *gaze* SP exceeds *ocular* auditory SP. Finally, SP of periodic (triangular) motion trajectories revealed that auditory, like visual, SP improves rapidly over time, indicating predictive behavior. In sum, auditory SP closely mimics visual SP but with reduced gain. We propose that auditory motion processing exists, is robust, and recruits a velocity-dependent neural process shared with vision.

9:25

1aPPa5. Perceived auditory motion is inaccurate during smooth head rotation. Tom C. Freeman, John F. Culling (Psych., Cardiff Univ., Tower Bldg., Park Pl., Cardiff, South Glamorgan CF10 3AT, United Kingdom, freemant@cardiff.ac.uk), Michael A. Akeroyd, and W Owen Brimijoin (Glasgow Royal Infirmary, MRC/CSO Inst. of Hearing Res. (Scottish Section), Glasgow, United Kingdom)

Hearing is confronted by a similar problem to vision as the observer moves. Movement of the sensors creates image motion that remains ambiguous until the observer knows the velocity of eye and head. The visual system solves this problem using motor commands, proprioception, and vestibular information (so-called "extra-retinal signals"), but the solution is not always perfect. Here, we compare the auditory errors made during head rotation with the visual mistakes made during eye movement. Real-time measurements of head velocity were used to change the gain relating head movement to source movement across a loudspeaker array. The gain at which "extra-cochlear signals" (encoding head rotation) was perceptually matched to "acoustic signals" (encoding source motion across the ears), thus yielding the perception of a stationary source, was small and positive. The gain varied depending on context, e.g., average source direction with respect to head. Two possible accounts of analogous findings in vision will be discussed, one based on differences in neural signal accuracy and the other based on Bayesian estimates that resolve differences in neural signal precision. We consider the degree to which these explanations can be applied to auditory motion perception in moving listeners.

9:45–10:00 Break

10:00

1aPPa6. From a nonuniform brain map to motion selectivity in the owl's midbrain. Jose L. Pena (Neurosci., Albert Einstein College of Medicine, 1410 Pelham Pkwy South, Kennedy Ctr. Rm, 529, Bronx, NY 10461, jose.pena@einstein.yu.edu), Brian J. Fischer (Mathematics, Seattle Univ., Seattle, WA), and Yoram Gutfreund (Physiol. and Biophys., Technion Med. School, Haifa, Israel)

The owl's midbrain displays a map of auditory space. This map is a distorted representation of the environment, where the front is magnified. In addition, sound is differently attenuated by the head depending on direction, where gain increases in the front. Because neurons in the map are functionally interconnected, the nonuniform representation influences the processing of features that rely on integration across space and time. In addition, the nonuniform gain deforms spatial receptive fields, affecting history-dependent responses. As a result, neurons become sensitive to motion direction. Previous work has explained the owl's localizing behavior by statistical inference, where uncertainty about the sensory input and prior information can be combined optimally to guide behavior. This theory can be applied to moving targets, where sensory cues must be integrated over time. This analysis shows that the midbrain neural population can be readout to predict future positions of moving targets, a critical function for a predator species. Thus, the nonuniform representation of space can induce biased computation of a higher-order stimulus feature, allowing systematic direction-selectivity and predictive power. Because neural representations where ethologically important ranges are overrepresented, are widespread in the brain, these mechanisms are likely observed in other sensory maps that guide behavior.

10:20

1aPPa7. Detection and tracking of fluttering moths by echolocating bats. Wu-Jung Lee and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., 3400 N Charles St., Ames Hall 132, Baltimore, MD 21218, wjlee@jhu.edu)

Aerial-hawking echolocating bats present an interesting model for studying how a moving listener interacts with moving sound sources. During foraging, relative movements between the bat and its prey introduce echo variation, which is further influenced by the timing of biosonar emissions with respect to insect wingbeats. Through systematic characterization of echoes from fluttering moths, this study aims at understanding how intermittent and highly variable echo information may be perceived and integrated by foraging bats for prey detection, tracking, and discrimination. Calibrated broadband echoes from frequency-modulated linear sonar sweeps were measured from live, fluttering moths of different morphological features with concurrent high-speed video recordings. The measurements are used to estimate the probability distribution of echo amplitudes as a function of moth morphology, wingbeat phase, and body orientation, as well as to predict changes of echo characteristics as the bat approaches the prey. Signal detection theory is then applied to model the bat's perception of prey presence as influenced by changes in biosonar emission rate in different phases of the foraging process. These results can be further integrated with neurophysiological findings on the bat's internal representation of space for a better understanding of this agile and efficient biosonar system.

10:40

1aPPa8. Navigating the world using echo flow patterns. Michaela Warnecke and Cynthia F. Moss (Dept. of Psych and Brain Sci., Johns Hopkins Univ., 3400 N Charles St., Baltimore, MD 21218, warnecke@jhu.edu)

As an animal moves through the environment, it experiences flow of sensory information from stationary objects. Animals that rely largely on visual information to guide their movement experience optic flow patterns as they navigate, which can be used to measure the relative distance of objects in the environment (Gibson, 1958, Besl, 1988). For example, honeybees use optic flow patterns to center themselves in a flight corridor, and experimental manipulations of the visual patterns on the walls directly influence the animal's navigation path (Srinivasan *et al.*, 1996). Other animals instead show wall-following behavior, choosing to navigate close to visual obstacles (Scholtyssek *et al.*, 2014). Here, we report on the navigation paths of animals that rely on acoustic signals to guide their movement. Echolocating bats emit ultrasonic signals that reflect from objects in the path of their sound beam, and we are studying how these animals use echo flow to guide their flight path through a corridor. In this study, we flew echolocating big brown bats through tunnels of horizontally and vertically hung PVC pipes to investigate how acoustic patterns influence the bat's flight and echolocation behavior.

11:00

1aPPa9. Aurally aided visual search performance for stationary and walking listeners. Douglas Brungart, Tricia Kwiatkowski (Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.brungart@us.army.mil), Sarah E. Kruger (National Intrepid Ctr. of Excellence, Bethesda, MD), Julie Cohen, Thomas A. Heil, and Danielle Zion (Walter Reed NMMC, Bethesda, MD)

One of the most important functions of spatial hearing is to facilitate the rapid shift of visual gaze in the direction of newly detected sound sources that might represent potential targets or threats in our environment. Some of the most dangerous threats can emerge when listeners are walking, so it is useful to know what impact walking might have on audiovisual target detection and identification in complex scenes. This study used a virtual environment consisting of a motorized treadmill surrounded by a 180^{o} projection screen mounted in front of a 64-loudspeaker array to evaluate how quickly participants were able to detect a visual target displayed in the presence of multiple distracters. The distracters were small clusters containing 2 or 4 dots, and the target was a small cluster containing 1 or 3 dots that was presented from the same location as a pulsed broadband noise. The task was to find the target as quickly as possible and press a button to indicate if it had 1 or 3 dots. Data were also collected in an auditory localization condition that required listeners to move a cursor to the location of the sound source and a visual-only condition that required participants to perform the visual search task with no auditory cue. The results show that target identification times were generally reduced when the listener was walking, suggesting that the increased motor activation caused by walking may enhance the ability to perform audiovisual searches. [Research Supported by DoD PHTBI award (W81XWH-12-2-0068).]

11:20

1aPPa10. On the role of visual information about head motion in the interpretation of dynamic interaural cues for front/back sound localization. Ewan A. Macpherson (National Ctr. for Audiol. & School of Communication Sci. and Disord., Western Univ., Elborn College, 2262, 1201 Western Rd., London, Ontario N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

The dynamic interaural cues generated by listener head rotation specify the front/back location of a sound source only when coupled with information about the head motion that produced them. Potential sources of this information are vestibular, proprioceptive, and visual inputs. To investigate the influence of these extra-auditory modalities on the interpretation of dynamic acoustic cues, we use real-time motion tracking and dynamic virtual auditory space synthesis in conjunction with an oscillating chair apparatus that permits dissociation of head-on-body and head-in-space motion. The results of a previous study conducted in darkness [Kim, Barnett-Cowan, & Macpherson, ICA 2013], suggest that vestibular (head-in-space) information is necessary and sufficient for accurate interpretation of dynamic cues, whereas proprioceptive (head-on-body) information is neither necessary nor sufficient. In the present study, active (providing vestibular and proprioceptive information) or passive (providing primarily vestibular information) head-in-space rotations at 50°/s were combined with congruent or incongruent visual motion information in a task requiring the identification of the front/back location of 200- or 400-ms low-pass noise targets. Experiments were conducted in a lighted room, and incongruent visual stimuli were produced with left/right-reversing prism glasses providing ~30° field of view. We found that this visual input had little or no influence on listeners' interpretation of the dynamic auditory cues.

11:40

1aPPa11. Changes in the integration of self motion and auditory spatial cues with age, hearing impairment, and use of hearing devices. W. O. Brimijoin and Micheal A. Akeroyd (Inst. of Hearing Res. - Scottish Section, MRC/CSO, MRC/CSO Inst. of Hearing Res., 10-16 Alexandra Parade, Glasgow G31 2ER, United Kingdom, owen@ihr.gla.ac.uk)

To resolve a front/back confusion, listeners may use both the spectral cues associated with the pinnae and they may turn their heads and note the direction in which the signal moves. Since hearing impairment typically involves the loss of high frequency information, one might expect that hearing impaired listeners would be more reliant on self-motion cues. Such listeners, however, often wear hearing aids that alter binaural level cues, suggesting they may distort dynamic self-motion-related cues. To examine these interactions, we utilized a previously published front/back illusion [W.O. Brimijoin and M.A. Akeroyd, iPercept. 3(3), 179–182 (2012)]: the perceptual location of signals whose position is moved at twice the angular rate of head movements is opposite to its physical location. In normal-hearing listeners, the illusion is powerful but weakens with increasing low-pass filter cutoff frequency. We found that for hearing-impaired listeners, the illusion was effective at all cutoff frequencies. The effect of hearing aids was heterogeneous across listeners, but in no case was a listener returned to normal performance, suggesting that hearing aids are not only failing to provide the listener with spatially-informative spectral cues, but they may interfere with self-motion-related cues. [Work supported by MRC (U135097131) and the Chief Scientist Office (Scotland).]

MONDAY MORNING, 18 MAY 2015 — BALLROOM 2, 8:00 A.M. TO 12:00 NOON

Session 1aPPb

Psychological and Physiological Acoustics: General Topics in Psychological Acoustics (Poster Session)

Nirmal Kumar Srinivasan, Chair
National Center for Rehabilitative Auditory Research, 3710 SW US Veterans Hospital Road, Portland, OR 97239

Posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 8:15 a.m. to 9:45 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m. There will be a 15-minute break from 9:45 a.m. to 10:00 a.m.

Contributed Papers

1aPPb1. The three-dimensional morphological database for spatial hearing research of the BiLi project. Felipe Rugeles Ospina (Orange Labs, 4 rue du Clos Courtel, Cesson Sevigné 35510, France, felipe.rugelesospina@orange.com), Marc Emerit (Orange Labs, Cesson Sévigné, France), and Brian F. Katz (LIMSI, Orsay, France)

The BiLi Project is a French collaborative project concerned with binaural listening and improved rendering for the general public. One of the goals of this project is to develop simple methods to personalize head-related transfer functions based on people's morphology. In order to accomplish this, it is necessary to study the links between an individual's HRTF and their corresponding morphology. As a resource for studies relating these parameters, two databases have been created: a database of high resolution measured HRTFs and an accompanying database of 3-D morphological data. This work presents the details of the creation of the morphological database. The prerequisites for such a database are presented. Various technical solutions are proposed and evaluated. Resulting accuracies of the methods are compared using extracted morphological parameters as defined in the CIPIC database with those measured directly on the individuals.

1aPPb2. On the importance of information-bearing acoustic changes for understanding speech in simulated electrical-acoustic stimulation. Christian Stilp (Dept. of Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu), Gail Donaldson, Soohee Oh (Univ. of South Florida, Tampa, FL), and Ying-Yee Kong (Northeastern Univ., Boston, MA)

Listeners utilize information-bearing acoustic changes in the speech signal to understand sentences. This has been demonstrated in full-spectrum speech using cochlea-scaled entropy (CSE; Stilp & Kluender, 2010 *PNAS*) and in vocoded speech (CSE_{CI}; Stilp *et al.*, 2013 *JASA*). In simulations of electrical-acoustic stimulation (EAS), vocoded speech intelligibility is aided by the preservation of low-frequency acoustic cues. The extent to which listeners rely on information-bearing acoustic changes to understand EAS speech is unclear. Here, normal-hearing listeners were presented noise-vocoded sentences with 3–6 spectral channels in two conditions: 1) vocoder-only (80–8000 Hz, filtered using third-order elliptical filters), and 2) simulated hybrid EAS (vocoded >500 Hz; original acoustic <500 Hz). In each sentence, four 80-ms intervals containing high-CSE_{CI} or low-CSE_{CI} acoustic changes were replaced with speech-shaped noise. As expected, performance improved with more channels and the preservation of low-frequency fine-structure cues (EAS). Relative to control vocoded sentences with no noise replacement, performance was impaired more when high-CSE_{CI} intervals were replaced by noise than when low-CSE_{CI} intervals were replaced in 5- and 6-channel sentences, but not at lower spectral resolutions. This effect maintained across vocoder-only and EAS sentences. Findings support the conclusion that EAS users make use of information-bearing acoustic changes to understand speech.

1aPPb3. A study on sound contents development based on analysis a Foley sound and a real sound of thunder. Ahn Iksoo (TeleCommun. & Information, soolsil Univ., 369 sangdo-ro, Dongjak-gu, Seoul 156-743, South Korea, aisbestman@naver.com), Bae Seong Geon (daelim, Anyang, South Korea), and Bae Myungjin (TeleCommun. & Information, soolsil Univ., Seoul, South Korea)

This study focuses on verifying possibility of developing Foley sound of thunder, one of the tools of Foley sound used to make sound effect required for story making of radio drama in early period broadcasting as a sound content by examining it. The purpose of this research is to make its creativity and uniqueness into sound content by proving similarities between thunder Foley sound made by tools and actual Foley sound based on their comparison and analysis and studying its production principles and usage.

1aPPb4. Predicting the timing of future events using sound: Bouncing balls and tone bursts. Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 600 S. Paulina Str., AAC 1012, Chicago, IL 60612, valeriy_shafiro@rush.edu), Brian Gygi (Speech and Hearing Res., U.S. Dept. of Veterans Affairs Northern California Health Care System, Martinez, CA), and Anatoliy Kharkhurin (Dept. of Int.. Studies, American Univ. of Sharjah, Sharjah, United Arab Emirates)

Listeners can predict the timing of the next bounce of a bouncing ball by sound alone with high accuracy, relying primarily on temporal cues [Giordano *et al.*, 2011, JASA, 129, 2594]. The present study investigated the role of temporal structure (natural bouncing patterns versus artificially reversed bouncing patterns) and event type (ball bouncing sounds versus similarly patterned tone bursts). After listening to two to four such sounds, listeners would indicate when they expected the next sound to occur (without hearing it). Results replicate previous findings of high accuracy in predicting the timing of natural bounce patterns regardless of event type. In contrast, accuracy was substantially poorer when pattern timing was reversed. Nevertheless, even for reversed patterns, listener accuracy improved as a greater number of bounces were heard prior to response time. This suggests that with additional information listeners were able to utilize veridical acoustic cues that best fit the temporal pattern. These findings demonstrate that in addition to being highly adept in estimating future timing of natural auditory events, listeners (a) tend to rely on temporal dynamics of everyday events in their timing estimates, and (b) can modify their response strategies for artificial timing patterns.

1aPPb5. An analysis the actual sound and Foley sound at stepping on dead leaves. Ahn Iksoo, Myungjin Bae, and Seonggeon Bae (TeleCommun. & Information, soolsil Univ., 369 sangdo-ro, Dongjak-gu, Seoul 156-743, South Korea, aisbestman@naver.com)

The purpose of this research is to prove similarity between imitation sound of stepping on dead leaves used in media and actual sound of stepping on dead leaves and based on the result applying it to sound contents. By comparing imitation and actual sound, this research proves that imitated sound and the tools used for this are useful as sound contents. Also, it concludes that imitation sound of stepping on dead leaves has big potential to be developed as sound contents for not only performance, exhibition, and experience contents but also for treating spiritual health of people.

1aPPb6. Temporal model of edge pitch effects. Peter Cariani, H. Steven Colburn (Hearing Res. Ctr., Boston Univ., 44 Cummington St., Boston, MA 02215, cariani@bu.edu), and William M. Hartmann (Phys. and Astronomy, Michigan State Univ., East Lansing, MI)

Musical pitches can be perceived in broadband noise stimuli near frequencies corresponding to parameter changes along the frequency dimension. For example, monaural edge pitches (MEPs) are produced by noise stimuli with sharp spectral edges. Binaural edge pitches (BEPs) are produced by dichotic noise with interaural-phase changes at frequency boundaries (e.g., 0-pi), and binaural-coherence edge pitches (BICEPs) arise from boundaries between frequency regions with different interaural coherence (such as changes from correlated to uncorrelated noises). Perceived pitches are shifted slightly in frequency away from spectral edges: MEPs are shifted into noise bands, BICEPs are shifted into the incoherent region, and BEPs are shifted bimodally. This presentation proposes a temporal model for these edge pitches based on population-wide all-order interspike interval distributions (summary autocorrelations, SACFs), computed using the Zilany-Bruce-Carney 2014 model of the auditory nerve. Pitches were estimated from mean SACF values among lags at all F0-subharmonics (0–30 ms). Binaural pitches were estimated from computations based on corresponding left and right PSTs after a binaural delay and cancellation process. Model predictions agreed well with pitch judgments in both monaural and binaural cases. [Work supported by NIDCD (R01 DC00100 and P30 DC004663) and by AFOSR FA9550-11-1-0101.]

1aPPb7. Relative weight of temporal envelopes across speech frequency regions for speech intelligibility in hearing-impaired listeners and cochlear implant users. Yingjiu Nie, Harley J. Wheeler, Alexandra B. Short, and Caleb W. Harrington (Commun. Sci. and Disord., James Madison Univ., 801 Carrier Dr. - MSC 4304, Harrisonburg, VA 22807, nieyx@jmu.edu)

The study was aimed to investigate, among three groups of listeners—normal-hearing, hearing-impaired, and cochlear implant users, the relative weight of temporal envelopes for speech intelligibility in each of the eight frequency regions ranging between 72 and 9200 Hz. Listeners were tested in quiet and in the presence of steady or amplitude modulated noise at two rates (4 and 16 Hz). An eight-band vocoder was implemented when testing the acoustic-hearing groups. Speech intelligibility of a given region/band was assessed by comparing scores in two conditions differing only by the presence or absence of the band of interest; the proportion of the derived score to the sum across the eight regions/bands was computed as the relative weight. Preliminary data showed the following: (1) in quiet, similar frequency-weighting pattern for all three groups with higher weight in the mid/mid-high frequency range; (2) in noise, for the normal-hearing group, different weighting patterns between steady noise and amplitude-modulated noise; for the other two groups, similar weighting patterns for all types of noise with comparable weight across bands # 2–7 and lower weight for the bands # 1 and 8. The contribution of each region/band to masking release will also be discussed.

1aPPb8. Analytic and divided listening in normal-hearing and hearing-impaired listeners measured in a nonspeech pattern identification task. Elin Roverud, Virginia Best, Christine R. Mason, and Gerald Kidd, Jr. (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, erover@bu.edu)

In multisource listening environments, it is important to be able to attend to a single sound source (analytic listening) while concurrently monitoring unattended sources for potentially useful information (divided listening). Previous studies have indicated that hearing-impaired (HI) listeners have more difficulty with analytic listening than do normal-hearing (NH) listeners. Although central factors (e.g., selective attention and memory) clearly play a role, the extent to which differences in peripheral factors (e.g., auditory filter characteristics) contribute to this effect is not clear. In this study, performance in a closed-set nonspeech tonal pattern identification task was measured in NH and HI listeners for patterns centered at 850 and 3500 Hz. The frequency spacing among the tones forming the patterns was adjusted to equate performance across listeners at each center frequency separately to control for peripheral frequency resolution. Patterns were then played at both frequencies concurrently. Listeners were instructed to attend to either the low or high frequency and identify the pattern. In a second condition, patterns were randomly presented at one frequency with a foil at the other frequency, requiring the listener to monitor both frequencies to identify the pattern. Preliminary findings suggest that peripheral and central factors contributed to performance. [Support: NIH-NIDCD.]

1aPPb9. Voice emotion recognition and production by individuals with normal hearing and with cochlear implants. Monita Chatterjee, Aditya M. Kulkarni, Julie A. Christensen (Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, monita.chatterjee@boystown.org), Mickael L. Deroche, and Charles J. Limb (Otolaryngol., Johns Hopkins Univ. School of Medicine, Baltimore, MD)

Children and adults with normal hearing (NH) as well as those who use cochlear implants (CIs) were tested on a voice emotion recognition task. The materials were child-directed sentences, and acoustic analyses showed that features such as voice pitch range were exaggerated relative to earlier reports in the literature with adult-directed materials. The NH participants achieved ceiling-level performance in the task, while the children and adults with CIs had lower, and highly variable, scores. In a parallel study, we have also collected complex pitch discrimination thresholds in many of these participants. In a new preliminary study, we are analyzing the acoustic features of voice emotion production by these populations. The task involves reading simple sentences in a happy and a sad way. In this presentation, we will report on relationships between the perceptual data on voice emotion recognition and complex pitch discrimination by child and adult NH and CI participants. In addition, we will report on our initial acoustic analyses of voice emotion production by participants with NH and those with CIs.

1aPPb10. Spectral resolution and speech recognition in noise for children with hearing loss. Ryan W. McCreery (Audiol., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, ryan.mccreery@boystown.org), Jenna M. Browning (Univ. of North Carolina, Chapel Hill, NC), Benjamin Kirby, Meredith Spratford, and Marc Brennan (Audiol., Boys Town National Res. Hospital, Omaha, NE)

Better spectral resolution has been associated with higher speech recognition in noise in adults with hearing loss who use hearing aids and adults with cochlear implants. However, the role of signal audibility and age on this relationship has not been reported. The goal of this study was to evaluate the effect of aided audibility and spectral resolution on speech recognition in noise for a group of children with sensorineural hearing loss and a group of adults with hearing loss. Higher age, better aided audibility, and the ability to detect more ripples per octave in a spectral ripple discrimination task were associated with better sentence recognition in noise for children with hearing loss.

1aPPb11. Application of the WKB method for an active cochlear model. Amir Nankali (Mech. Eng., Univ. of Michigan, 1092 Louise St., Apt. 14, Ypsilanti, MI 48197, nankali@umich.edu)

The Wentzel-Kramers-Brillouin (WKB) method has been used to approximate the solution for systems with slowly varying properties. The analytic method is computationally efficient and provides insights into the physical problem by decomposing the solution into a dominant wavenumber and the associated amplitude [e.g., Steele and Taber (1978)]. Because of its computational efficiency, this method provides a convenient means to estimate parameters in a complicated cochlear model, through variation and optimization. In turn, these parameters can be used in a more complete approximation technique (like a finite element or boundary element method) where parameter estimation can be more cumbersome. In this paper, we extend the WKB approximation to an active cochlear model, including the micromechanics of the organ of Corti (OoC) and electromotility. The model involves the OoC structural elements, basilar membrane (BM), and tectorial membrane (TM), coupled to the electrical potentials in the cochlear ducts and fluid pressure. Model predictions are compared to numerical approximations using finite elements. [Work supported by NIH-NIDCD R01-04084 and NIH NIDCD-T32-000011.]

1aPPb12. Priming gestures with associated sounds. Guillaume Lemaitre (Dept. of Psych., Carnegie Mellon Univ., IRCAM, 1 Pl. stravinsky, Paris 75004, France, GuillaumeJLemaitre@gmail.com), Nicole Navolio, and Laurie M. Heller (Dept. of Psych., Carnegie Mellon Univ., Pittsburgh, PA)

Hommel (1996) established that sounds artificially associated with key presses could prime key presses. The goal of the current study was to explore the nature of the link between auditory perception and manual actions by comparing the priming of manual actions by naturally and artificially associated sounds. We report three experiments. The procedure in each experiment consisted of cueing participants to perform manual gestures. Executing the gestures produced feedback sounds; those sounds were also used as primes before the response cues. Experiment One replicated Hommel's procedure: participants lifted keys that triggered artificial sounds. Experiment Two used sounds naturally produced by tapping or scraping wooden dowels on a custom interface. Experiment Three replicated Experiment Two with the sounds of the dowels muffled. The priming effect was observed on reaction times in Experiments One and Two but not in Experiment Three. These results show that long-term associations between sounds and gestures created in memory by repeated experience throughout life are not sufficient to prime the gestures. Instead they suggest that auditory-motor priming may be mediated by associations between sounds and gestures that are created on-line by the experiment via associative learning, and which are short-lived and can be readily reconfigured.

1aPPb13. Maximum equivalent pressure output and maximum stable gain of a light-activated contact hearing device that mechanically stimulates the umbo: Temporal bone measurements. Sunil Puria (EarLens Corp., 496 Lomita Mall, Stanford, CA 94305, puria@stanford.edu), Rodney Perkins (EarLens Corp., Menlo Park, CA), and Peter Santa Maria (Otolaryngology-HNS, Stanford Univ., Stanford, CA)

The non-surgical contact hearing device (CHD) consists of a light-activated balanced-armature tympanic-membrane transducer (TMT) that drives the middle ear through contact with the umbo, and a behind-the-ear unit with a widely vented light-emitter Assembly inserted into the ear canal that encodes amplified sound into pulses of light that drive and power the TMT. In comparison to acoustic hearing aids, the CHD is designed to provide higher levels of maximum equivalent pressure output (MEPO) over a broader frequency range and a significantly higher maximum stable gain (MSG). No artificial middle-ear model yet exists for testing the CHD, so we instead measured it using fresh human cadaveric temporal bones. To calculate the MEPO and MSG, we measured the pressure close to the eardrum and stapes velocity for sound drive and light drive using the CHD. The baseline sound-driven measurements are consistent with previous reports in temporal bones. The average MEPO ($N = 4$) varies from 116 to 128 dB SPL in the 0.7 to 10 kHz range, with the peak occurring at 7.6 kHz. From 0.1–0.7 kHz, it varies from 83 to 121 dB SPL. For the average MSG, a minimum of about 10 dB occurs in the 1–4 kHz range, above which it rises as high as 42 dB at 7.6 kHz. From 0.2 to 1 kHz, the MSG decreases linearly from about 40 dB to 10 dB. The measurements of MEPO and MSG are compared with predictions from circuit-model calculations. The CHD may offer a way of providing broad-spectrum amplification appropriate to treat listeners with mild-to-severe hearing impairment.

1aPPb14. Evaluation of a navigational application using auditory feedback to avoid veering for blind users on Android platform. György Wersényi (Dept. of Telecommunications, Széchenyi István Univ., Egyetem tér 1, Györ 9026, Hungary, wersenyi@sze.hu)

Assistive technologies incorporate various applications and devices to assist handicapped people. This includes areas such as user interface design, sound design, mapping of visual and spatial information, sonification, etc. Visually impaired people also use state-of-the-art solutions especially for safe and independent navigation with the help of electronic travel aids (ETAs). Smartphones and tablets offer new possibilities in creating applications using the in-built sensors and open-source development platforms such as Android. This paper presents a short overview of recent technical approaches to assist users with visual impairment focusing on mobile applications for the Android platform. Evaluation of a navigational assistant using the magnetic sensor and tactile/auditory feedback to avoid veering is presented. Users can use the application after a short training time and keep straight walking path in outdoor and indoor tests using different sounds and vibration of the smartphone.

1aPPb15. Effect of some basic and premium hearing aid technologies on non-speech sound acceptability. Jingjing Xu, Jani A. Johnson, and Robyn M. Cox (School of Commun. Sci. and Disord., Univ. of Memphis, 807 Jefferson Ave., Memphis, TN 38105, jxu3@memphis.edu)

Acceptability of everyday non-speech sounds is closely related to hearing aid (HA) satisfaction. Acceptability is determined by a listener's overall impression of a sound when its different aspects, such as loudness, naturalness, and clarity, are considered. Various HA features, especially digital noise reduction (DNR), are designed to improve acceptability. Compared to basic HAs, premium HAs have more advanced DNR functions, as well as other unique features that are not included in basic HAs. Manufacturers often claim that everyday non-speech sounds are more acceptable when listening with premium HAs relative to basic HAs. However, there is minimal evidence to support this claim. This study evaluated acceptability of non-speech sounds in laboratory and real-world settings when using exemplars of basic and premium HAs. Forty-five older adults with mild-to-moderate hearing loss were bilaterally fitted with four pairs of BTEs (two basic and two premium) from two major manufacturers. Outcomes were obtained for each pair after a four-week field trial. Laboratory data were acceptability ratings of 21 real-time produced everyday sounds with different durations and intensities. Self-report data were rating scores from three questionnaires. No evidence was found in this study to show that premium HAs yield greater acceptability than basic. (Work supported by NIDCD.)

1aPPb16. Effect of spectral amplitude elevation for vowel identification in simulated combined electric-and-acoustic hearing. Fei Chen (Dept. of Elec. and Electron. Eng., South Univ. of Sci. and Technol. of China, Shenzhen, Guangdong 518055, China, fchen@sustc.edu.cn)

Though the benefits of combined electric-and-acoustic stimulation (EAS) have been widely reported, its underlying mechanisms responsible are not well understood yet. This study assessed the effect of balanced spectral amplitudes between acoustic and electric portions in EAS to vowel identification via two experiments. Four vowels (i.e., /iy/, /eh/, /oo/ and /ah/) were synthesized using pure-tone harmonic components. In Experiment 1, the spectral amplitudes of natural vowels were elevated for either acoustic (below 600 Hz) or electric portions (above 600 Hz) from 5 dB to 30 dB. Vowel identification scores were collected from eight normal-hearing listeners. In Experiment 2, the synthesized vowel stimuli were processed by the signal processing condition simulating the electric-and-acoustic stimulation. The results of identifying synthesized vowels and synthesized EAS-processed vowels in the two experiments showed the similar patterns of declined identification rates as a function of (acoustic or electric) spectral amplitude elevation. Furthermore, the specific loudness pattern computed from Moore *et al.*'s model was used to predict the vowel identification score, yielding a good prediction performance (i.e., $r = 0.9$). The results in the present work suggested the importance to maintain the balance between the spectral amplitudes of acoustic and electric portions for vowel identification in EAS hearing.

1aPPb17. Speech experience and cue trading in budgerigars. Mary M. Flaherty (Psych., SUNY Buffalo, 392 Park Hall, North Campus, Buffalo, NY 14260, maryflah@buffalo.edu), James R. Sawusch, and Micheal L. Dent (Psych., SUNY Buffalo, Amherst, NY)

The current project investigates how experience with human speech can influence speech perception in budgerigars. Budgerigars are vocal mimics and speech exposure can be tightly controlled in a laboratory setting. The data collected include behavioral responses from 30 budgerigars, tested using a cue-trading paradigm with synthetic speech stimuli. Prior to testing, the birds were divided into three exposure groups: Passive speech exposure (regular exposure to human speech), no speech exposure (completely isolated), and speech-trained (using the Model-Rival Method). After the exposure period, all budgerigars were tested using operant conditioning procedures. Birds were trained to peck keys in response to hearing different synthetic speech sounds that began with either "d" or "t." Sounds varied in VOT and in the frequency of the first formant. Once training performance reached 80% on the series endpoints, budgerigars were presented with the entire series, including ambiguous sounds. The responses on these trials were used to determine which speech cues the birds use, if cue trading behavior was present, and whether speech exposure had an influence on perception. Preliminary data suggest experience with speech sounds is not necessary for cue trading by budgerigars.

1aPPb18. Can you say that again? Aging and repetition effects in different types of maskers. Karen S. Helfer, Richard L. Freyman, Angela Costanzi, Sarah Laakso, and Gabrielle Merchant (Commun. Disord., Univ. of Massachusetts Amherst, 358 N. Pleasant St., Amherst, MA 01002, khelfer@comdis.umass.edu)

The most common strategy used by listeners when they cannot understand an utterance is a request for repetition. Repetition may improve perception in a number of ways, including priming the target message via sharpening the representation of aspects of the signal centrally, buying additional time for the listener to process the message, and enhancing the coding of information into memory. In the present study, the benefit of hearing an utterance a second time was measured in three different types of maskers (competing speech, single-channel envelope modulated noise, and steady-state speech shaped noise) using two different tasks: immediate recognition of the target sentence and memory for words presented during a previous block of trials. For the latter task, participants were presented with isolated words and were asked if those words appeared in the previous set of sentences. Participants (younger, middle-aged, and older adults) also completed a battery of cognitive tasks that included measures of processing speed, attentional control, and working memory. This presentation will describe results of analyses comparing the benefit of repetition for immediate recall and memory among the listener groups in the presence of different types of maskers. (Work supported by NIDCD R01 DC012057.)

1aPPb19. Listener consistency in identifying speech mixed with particular "bubble" noise instances. Michael I. Mandel, Sarah E. Yoho, and Eric W. Healy (Speech and Hearing Sci., The Ohio State Univ., 395 Dreese Labs, 2015 Neil Ave., Columbus, OH 43210, mandelm@cse.ohio-state.edu)

Previous work has shown that the intelligibility of mixtures of the same exact speech token with different instances of "bubble" noise is highly dependent on the configuration of the random time-frequency glimpses it provides. In the current study, the consistency of these judgments was measured for such mixtures involving six consonants, /t/, /d/, /f/, /v/, /t͡ʃ/, /d͡ʒ/, in an /a/-consonant-/a/ context. Intra-subject consistency on ten repetitions each of 60 mixtures was found to be high, as was inter-subject consistency between five normal-hearing subjects on 1200 mixtures. In addition, the noise level at a small subset of time-frequency points was significantly correlated with the overall intelligibility of the mixtures of a given token, suggesting that these regions contribute more strongly to correct identification than others. The current study finds that these regions are quite consistent across the five subjects. For example, for the token /ada/, these points surround the stop burst and the resolved harmonics following the onset of voicing. These results show the promise of the "bubbles" methodology for identifying time-frequency regions of individual utterances that contribute most to their intelligibility, permitting future study of such regions for different types of speech and listeners. [Work supported by NIH.]

1aPPb20. The effect of amplitude envelope on spatial ventriloquism. Dominique Beauregard Cazabon (Psych., Neurosci. & Behaviour, McMaster Univ., 424 Togo Salmon Hall, 1280 Main St. West, Hamilton, Ontario L8S 4M2, Canada, beaureda@mcmaster.ca) and Michael Schutz (School of the Arts, McMaster Univ., Hamilton, Ontario, Canada)

Spatial ventriloquism occurs when a visual event and an auditory event happen simultaneously; location judgments for the sound source become biased toward the location of the visual event. Evidence suggests the brain attributes greater weight to spatial information provided by the visual stimulus because the visual system offers better spatial acuity; when the visual stimulus is deteriorated, the visual bias is reduced. Thus, the brain performs optimal bimodal integration: greater weight is given to the modality which provides more information. The present study aims to determine whether the amplitude envelope of sounds provides spatial localization information to the perceptual system. We used a psychophysical staircase procedure to measure spatial ventriloquism experienced by participants for sounds with percussive, flat, and time-reversed percussive envelopes. We hypothesize that percussive and reverse-percussive sounds provide more information and thus better sound localization acuity than flat sounds, which would result in smaller degrees of spatial ventriloquism for the former than the latter. The results yield insight into the brain's use of auditory cues in audio-visual integration.

1aPPb21. Role of high-frequency component in source segregation in multi-talkers condition. Jing Mi and H. Steven Colburn (Biomedical Eng., Boston Univ., 44 Cummington Mall, Rm. 427, Boston, MA 02215, jingmi@bu.edu)

This study explores the role of high-frequency components in generating spatial release from masking (SRM) in multiple-talker conditions. From a frequency-analysis perspective, one can fully separate multiple speech signals when their spectrograms do not overlap. The sparsity of speech can be related to spectral overlap and is explored here in the context of SRM. The interaural coherence coefficient (IACC), which is defined as the value of the cross-correlation function at the expected interaural time difference (ITD) location, can be a good indicator of sparsity for binaural signals. The sparser the stimuli in the frequency domain, the less overlap in spectrograms, which leads to higher short-term IACCs. In this study, a sound-segregation system based on binaural cues was used to separate multiple sound signals in a mixture. Then, the short-term IACC was calculated for each time-frequency slice in the segregated sound. It was shown that average IACCs for multi-speech conditions are higher than for multi-white-noise conditions, especially in high-frequency regions. Based on this result, it was hypothesized that high-frequency components, which are sparser than low-frequency components in speech, are critical for SRM in multiple-talker conditions. Psychoacoustic evidence was found to support this hypothesis. [Work supported by NIH/NIDCD: Grant R01 DC00100.]

1aPPb22. The acoustic survey of intonation in Autism Spectrum Disorder. Zahra Azizi, (Linguist, Ferdowsi Univ. of Mashad, Modarres Boulevard – Dasht-e Chenar St. – Alley Number 7 – No 260, Shiraz 7158616976, Iran, zahra.azizima84@gmail.com)

The speech pattern of children with autism spectrum disorder (ASD) demonstrates a number of abnormal features, the study which can shed light on the difficulties the autistic children encounter in making meaning through speech. With regard to this stated purpose, in this article, I have scientifically examined the pattern of speech in a set of indicative and interrogative sentences as uttered by a group of autistic children and compared and contrasted the results with the pattern of speech in the same set of sentences as uttered by a group of typical development (TD) children. The specific aim of this study was to assess the intonation pattern, mean of pitch, amplitude, duration, intensity, and tilt in the ASD as compared to the TD. The data collected showed that while the amplitude in interrogative sentences, duration, intensity, mean of pitch, and tilt in ASD and TD were almost similar, the intonation pattern, the mean of pitch (with regard to severity of autism), and the amplitude in indicative sentences proved significantly different. To be more precise, with respect to these three latter features, the autistic children demonstrated monotony, an impairment that made them have difficulty making meaningful indicative and interrogative sentences.

1aPPb23. Envelope-power based prediction of auditory masking and speech intelligibility. Thomas Biberger and Stephan D. Ewert (Medizinische Physik and Cluster of Excellence Hearing4all, Carl-von-Ossietzky-Straße 9-11, Oldenburg, Lower Saxony 26135, Germany, thomas.biberger@uni-oldenburg.de)

In natural surroundings, people are often part of challenging acoustical scenarios (noise) interferers and reverberation. Amplitude modulations (AM) are fundamental acoustic features used to extract information in such situations. It has been shown that the envelope power spectrum model [EPSM; Ewert and Dau, J. Acoust. Soc. Am. **108**, 1181 (2000)] can account for psychoacoustic AM detection and masking data by considering the long-term envelope signal-to-noise-ratios (SNR) at the output of modulation filters. Recently, this concept was extended to consider short-term envelope SNRs to predict speech intelligibility in fluctuating maskers and in reverberation [mr-sEPSM; Jørgensen *et al*., J. Acoust. Soc. Am. **134**, 1475 (2013)]. In this study, the EPSM was further extended to consider envelope power SNRs (AM cues) on various time scales and "classic" power SNRs (intensity cues). The goal was to develop an auditory model for jointly predicting psychoacoustic masking and speech intelligibility. The model was demonstrated to account for a broad variety of psychoacoustic effects as AM detection, AM discrimination, AM masking, spectral masking, forward masking, intensity just-noticeable-differences (JNDs) and hearing threshold. Speech intelligibility was predicted comparable to Jørgensen *et al*.. Implications for future modeling including applications to audio and speech quality assessment are discussed.

1aPPb24. Individual differences in auditory brainstem response wave-V latency in forward masking: A measure of auditory neuropathy? Golbarg Mehraei (Health Sci. and Technol., Massachusetts Inst. of Technol., 23 Elm St. #5, Cambridge, MA 02139, gmehraei@gmail.com), Andreu P. Gallardo, Bastian Epp (Elec. Eng., Denmark Tech. Univ., Kgs. Lyngby, Denmark), Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA), and Torsten Dau (Elec. Eng., Denmark Tech. Univ., Kgs. Lyngby, Denmark)

A recent animal study suggests that noise exposure causes a preferential loss of low-spontaneous rate (low-SR) auditory nerve fibers (ANFs). This loss may leave detection thresholds normal yet degrade temporal encoding of supra-threshold sounds. Differences in the rate of recovery from forward masking in ANFs with different spontaneous rates may allow one to assess the state of different ANF populations. To test this, we measured auditory brainstem response (ABR) in a forward masking paradigm and evaluated wave-V latency changes with increasing masker-to-probe intervals (MPI). We expected that (1) loss of ANFs increases wave-V latency and forward masking thresholds, and (2) a preferential loss of low-SR fibers results in faster recovery time of wave-V latency. To test our hypotheses, we presented listeners with a broadband noise masker at two levels followed by a chirp probe at various MPIs. Initial results show that normal hearing threshold (NHT) listeners with delayed wave-V latency exhibit higher behavioral detection thresholds. Additionally, the listeners with the poorest behavioral thresholds had the fastest threshold recovery as a function of MPI. These results are consistent with the hypothesis that a preferential loss of low-SR fibers explains differences in NHT listeners.

1aPPb25. Masked speech recognition in a single-talker or a single-talker-plus-noise masker in school-age children and adults. Heather Porter (Otolaryngology/Head and Neck Surgery, The Univ. of North Carolina at Chapel Hill, 170 Manning Dr., CB 7070, Chapel Hill, NC 27599-7070, heather_porter@med.unc.edu), Lori J. Leibold (Allied Health, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Emily Buss (Otolaryngology/Head and Neck Surgery, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Previous work has shown that masked speech perception is poorer in younger children than older children and adults, particularly when the masker is composed of speech. The present study tested the hypothesis that young children's poor performance in speech maskers is due to reduced ability to benefit from transient improvements in the target-to-masker ratio (TMR). Listeners were normal-hearing school-age children (5–15 years) and adults. The target words and masker speech were produced by two different female talkers. Target stimuli were disyllabic words, and the masker was either: (1) speech-only or (2) speech-plus-noise. Both maskers included a continuous 60-dB-SPL stream of speech. The speech-plus-noise masker also included a continuous 50-dB-SPL speech-shaped noise. A four-alternative forced choice task was used, and target level was adapted to estimate threshold. Thresholds were higher for younger children than older children or adults for both masker conditions. While the inclusion of speech-shaped noise increased thresholds for all listeners, it had a smaller detrimental effect on younger children than on older children and adults. This result is consistent with the idea that greater susceptibility to speech-on-speech masking in younger children is related to a reduced ability to benefit from transient improvements in TMR.

1aPPb26. Dual-carrier vocoder: Effect on streaming in a cochlear implant simulation. Frederic Apoux, Brittney Carter, and Eric W. Healy (The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, fred.apoux@gmail.com)

Recently, it has been suggested that temporal fine structure (TFS) cues play a critical role in streaming while providing little speech information. To take advantage of this role of TFS, a coding strategy for cochlear implants (CIs) has been developed involving two independent carriers, one for the target and one for the background. The so-called dual-carrier coding can help to provide awareness of the background sounds to CI users while improving speech intelligibility. Two experiments evaluated the ability of normal-hearing (NH) listeners to process a target sentence and a "competing background" sentence and their ability to switch from one sentence to the other when presented with dual-carrier stimuli. The first experiment showed that NH listeners can deliberately focus their attention and successfully process the target or the background sentence. It also showed that this capability is not greatly affected by the addition of a 500 ms sentence fringe at the beginning of one of the sentences. Similarly, the second experiment showed that NH listeners could understand both sentences when presented with the same stimulus more than once. This last result confirms, if needed, that the target and background signals are both intelligible in dual-carrier stimuli. [Work supported by NIH.]

1aPPb27. An algorithm that generalizes to novel noise segments to improve speech intelligibility for hearing-impaired listeners. Eric W. Healy, Sarah E. Yoho (Speech & Hearing Sci., The Ohio State Univ., Pressey Hall Rm. 110, 1070 Carmack Rd., Columbus, OH 43210, healy.66@osu.edu), Jitong Chen, Yuxuan Wang, and DeLiang Wang (Comput. Sci. and Eng., The Ohio State Univ., Columbus, OH)

Machine-learning algorithms to extract speech from background noise hold considerable promise for alleviating limitations associated with hearing impairment. One of the most important considerations for implementing these algorithms into devices such as hearing aids and cochlear implants involves their ability to generalize to conditions not employed during the training stage. A major challenge involves the generalization to novel noise segments. In the current study, sentences were extracted from multi-talker babble and from cafeteria noise using an algorithm that estimates the ideal ratio mask by employing deep neural networks. Importantly, the algorithm was trained on segments of noise and tested using entirely different segments of the same nonstationary noise type. The training set was expanded through a noise-perturbation technique. Substantial benefit was observed for hearing-impaired listeners in both noise types, despite the use of unseen noise segments during the operational stage. Interestingly, normal-hearing listeners displayed benefit in babble but not in cafeteria noise. These results highlight the importance of evaluating these algorithms not only in human subjects, but in members of the actual target population. [Work supported by NIH.]

1aPPb28. Does providing more processing time improve speech intelligibility in hearing-impaired listeners? Virginia Best, Christine R. Mason, Jayaganesh Swaminathan, Elin Roverud, and Gerald Kidd (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, ginbest@bu.edu)

There is ample evidence in the literature that hearing loss increases the time taken to process speech (e.g., increased response times for word discrimination, sentence identification, and passage comprehension). This has led to the assumption that providing hearing-impaired listeners with more time to process speech would be beneficial. For sentence identification, a number of studies have examined the effects of adding silent gaps, or using time-expansion, but the results have been mixed. This may be because the distortion of natural sentence structure counteracted any benefits, or because the tasks used were not sufficiently demanding. In the current study, we re-examined this issue in young hearing-impaired listeners. By using sequences of words spoken in isolation, we were able to insert pauses between words without disrupting natural word boundaries. Furthermore, we used speech tasks involving interference in order to reveal effects that might only be apparent when the processing load is high. Although we found a statistically significant benefit of providing silent gaps between words, it was modest and was only present for some listeners in certain conditions. Overall, there was little evidence that the benefit was larger for hearing-impaired listeners or under more challenging conditions. [Work supported by NIH/NIDCD.]

1aPPb29. Speaking while listening: Broadbent revisited. Nandini Iyer, Eric Thompson, Brian Simpson, and Griffin Romigh (Air Force Res. Lab., 2610 Seventh St., BLDG 441, Wpafb, OH 45433, Nandini.Iyer.2@us.af.mil)

Air Force operators are often required to listen to several channels of ongoing radio communications while simultaneously responding to some or all of these messages. Broadbent (1952) showed that performance in such speaking-while-listening tasks typically resulted in a significant degradation. However, it was not clear if this decrement in performance was due to a competition of the resources required for simultaneous message comprehension and response formulation, or some other factor influencing the representation (such as poor signal quality). A replication of Broadbent's study was undertaken to measure the ability of subjects to perform in speaking-while-listening tasks. Listeners responded to a series of yes-no queries posed by a recorded talker regarding the presence/location of items on a visual display. Responses were scored based on a subject's ability to (1) respond to their assigned call-sign, (2) use the correct querier call sign in their response, and (3) provide a response to the question about the query item. Two variables were manipulated: rate of incoming messages and fidelity of the recordings (additive noise or radio communications). Results indicate that changing the rate of incoming messages had the largest impact on accuracy of responses, while the fidelity of the message had a relatively minor impact on performance, suggesting that limitations in these tasks may be largely due to an interference with response formulation.

1aPPb30. The role of age and executive function in auditory category learning. Rachel Reetzke (Commun. Sci. and Disord., Univ. of Texas at Austin, 2504A Whitis Ave., Austin, TX 78712, rreetzke@gmail.com), Todd Maddox (Psych., Univ. of Texas at Austin, Austin, TX), and Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX)

Auditory categorization is a natural and adaptive process that allows for the organization of high-dimensional, continuous acoustic information into discrete representations. The aim of this study is two-fold: (a) to examine the developmental trajectory of rule-based auditory category learning from childhood through early adulthood, and (b) to examine the extent to which individual differences in rule-based category learning relates to individual differences in executive function. Sixty participants with normal-hearing, 20 children (age

range, 7–12), 20 adolescents (age range, 13–19), and 20 young adults (age range, 20–23), learned to categorize novel spectrotemporally modulated sounds using trial-by-trial feedback. The experimental design included six blocks of 100 stimuli for a total of 600 trials. Results revealed that auditory categorization accuracy improved with age, with young adults outperforming children and adolescents. Computational modeling analyses indicated that the use of the task-optimal strategy (i.e., multidimensional learning strategy) improved with age. Further, correlational analyses revealed that the use of optimal multidimensional learning strategies was strongly correlated with individual differences in executive function, even when age was partialed out. The current findings demonstrate the protracted development of rule-based auditory categorization. The results further suggest that executive function strongly relates to successful strategy use during auditory category learning.

1aPPb31. Effect of age and hearing loss on intelligibility enhancement in a loudspeaker-based simulated reverberant environment. Nirmal Kumar Srinivasan (National Ctr. for Rehabilitative Auditory Res., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, nirmal.srinivasan@va.gov), Pavel Zahorik (Univ. of Louisville, Louisville, KY), Kasey M. Jakien, Samuel Gordon, Sean D. Kampel, Megan Stansell, and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland, OR)

The ability of the human auditory system to adapt to changes in the modulation spectrum of a reverberant signal has been documented (Zahorik *et al.*, 2012). Previously, results were presented examining the effects of age and hearing loss on adaptation to reverberation in an anechoic chamber with spatially separated speech with monaural distortions only. The results indicated that older listeners obtained decreased benefit from prior exposure to listening environments. However, in a realistic reverberant environment, both monaural distortions and spatial cues are present and it is important to analyze the effects of reverberation in the presence of spatial cues. In this experiment, a more spatially realistic sound field using 24 speakers separated by 15° in azimuth was created in an anechoic chamber by calculating the directions, attenuations, and delays of the reflections from a sound source. An impulse response for each loudspeaker that has the appropriate delays and attenuations for all reflections in its spatial vicinity was created. Speech was convolved with individual speakers' impulse responses for different spatial separations between target and maskers before presenting to the listeners. Effects of age and hearing loss on intelligibility enhancement due to prior exposure to a realistic listening environment will be discussed. [Work supported by NIH R01 DC011828.]

MONDAY MORNING, 18 MAY 2015 KINGS 4, 9:00 A.M. TO 12:00 NOON

Session 1aSC

Speech Communication and Psychological and Physiological Acoustics: Listening Effort I

Alexander L. Francis, Cochair
Purdue University, SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907

Christian Fullgrabe, Cochair
Institute of Hearing Research, Medical Research Council, Science Road, Nottingham NG7 2RD, United Kingdom

Chair's Introduction—9:00

Invited Papers

9:05

1aSC1. Research on listening effort: History and methods, theory, and practice. Alexander L. Francis (Purdue Univ., Speech, Lang. and Hearing Sci., Lyles-Porter Hall, 715 Clinic Dr., West Lafayette, IN 47907, francisa@purdue.edu) and Christian Füllgrabe (MRC Inst. of Hearing Res., Nottingham, United Kingdom)

The notion of listening effort has enjoyed a resurgence in recent years, with new work appearing in both theoretically and clinically oriented venues, and researchers pursuing this topic with an increasingly inventive suite of methods. Listening effort is often associated with perceptual and/or cognitive processing demands for understanding speech, terms that are often (but not necessarily) identified with mechanisms such as attention and working memory. Much of the recent research involving listening effort has arisen in the context of studies of people with hearing impairment and of people listening in adverse or sub-optimal conditions, thus highlighting a particular need to distinguish between measures of effort and more traditional measures of listening performance such as intelligibility. However, there is considerable terminological variety within the field, and significant debate concerning both the theoretical characterization and the clinical implications of all of these concepts. The goal of this talk is to provide a broad introduction to current research on listening effort, situating it within the context of a longer history of psychological studies of human performance and task demands and surveying some of the more common behavioral and psychophysiological methods used to assess listening effort for research and clinical purposes.

9:25

1aSC2. Listening effort and the division of auditory processing resources. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov)

When a listener is asked to simultaneously perform two different tasks, the results can reflect a wide range of factors. This presentation will describe data showing that performance reflects not only the division of a single resource shared between the tasks, but also the structure of the tasks themselves. Thus, if two tasks share structural resources (such as the same auditory pathway or the same memory representations) there may be interference even if there is still spare capacity at the level of other cognitive resources. Similarly, reduced effort may reflect independent structural constraints rather than increased capacity. For example, if the structure of the tasks allows the listener to rely upon a memory representation for one or both tasks, then a task in which the stimuli are presented simultaneously can be processed serially by relying on the memory trace to do the secondary task. The implications of this framework for studies examining listening effort will be discussed, along with ideas about ways to improve the design of experiments seeking to probe the capacity of a single shared resource.

9:45

1aSC3. When a competing talker is easy to ignore: The elusive nature of informational interference. Sven Mattys (Dept. of Psych., Univ. of York, York YO10 5DD, United Kingdom, sven.mattys@york.ac.uk)

A competing talker can impair speech processing through both energetic masking and informational, cognitive aspects of the competing utterance. We refer to the latter as informational interference. We hypothesized that handling informational interference depletes processing resources that could otherwise be allocated to recognizing the target speech. As a consequence, informational interference should be particularly pronounced for target sentences with high processing demands (syntactically complex sentences) than for sentences with low-processing demands (syntactically simple sentences). Using a speeded picture selection task, we assessed native and non-native listeners' understanding of subject-relative (simple) and object-relative (complex) sentences played against a competing talker vs. a matched energetic mask. While object-relative sentences were more difficult to process than subject-relative sentences, there was no effect of masker type, and this pattern was comparable for native and non-native listeners and across various SNRs. Thus, contrary to prior research, we found no evidence that a competing talker requires greater processing resources than energetic masking alone. Ongoing eye-tracking and pupillometric versions of this experiment will establish the nature of the discrepancy and the conditions under which informational interference is absent.

10:05

1aSC4. Downstream effects of accented speech on memory. Kristin Van Engen (Washington Univ. in St. Louis, One Brookings Dr., Campus Box 1125, Saint Louis, MO 63130-4899, kvanengen@wustl.edu)

When speech is acoustically challenging, listeners frequently engage additional cognitive processes to effectively perceive it. This increased listening effort can have downstream effects on behavior, impairing listeners' memory for what they have heard. To date, the listening effort literature has focused on processing speech in challenging listening situations that involve signal degradation due to noise and/or hearing impairment. A common theme in this literature is that acoustic degradation results in speech signals that deviate from listeners' stored phonological and lexical representations—a characteristic that degraded speech shares with speech produced in an unfamiliar accent. In this talk, we bring accented speech into the broader conversation about listening effort. In particular, we present preliminary results investigating the effects of intelligible foreign-accented speech on listeners' subsequent memory for words and sentences, even when these stimuli are intelligible. This work allows us to begin to incorporate listener effort associated with accented and degraded speech into a unified cognitive framework.

10:25–10:40 Break

10:40

1aSC5. The relationship between phonetic cue weighting and listening effort in listeners with cochlear implants. Matthew Winn, Jan Edwards, and Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Rm. 565, Madison, WI 53705, mwinn83@gmail.com)

In the study of speech perception, measures of cognitive load are especially useful because speech intelligibility performance results from a wide variety of auditory and cognitive processes that might demand different amounts of effort. The experience of elevated effort is an important part of hearing impairment and could be the target of audiological outcome measures. Listeners demonstrate different cue-weighting strategies when identifying phonemes, which is especially apparent for cochlear implant patients, who receive a highly degraded signal. This study explored whether changes in signal processing that affect phonetic cue weighting are also associated with changes in listening effort. Using both a conventional and an experimental speech processing strategy in alternating blocks, listeners completed tests of basic intelligibility, cue weighting for the /b/-/d/ contrast, and listening effort indicated by pupil dilation. Listeners for whom the experimental strategy promoted increased weighting of formant transition cues generally also showed reduced effort when using that strategy to listen to complete sentences, compared to listening with their basic strategy. Intelligibility scores generally did not show much change across both strategies. Results suggest that intelligibility measures can be complemented by other measures that tap into the mechanisms of perception and the effort demanded by the process.

11:00

1aSC6. Factors that increase processing demands when listening to speech. Ingrid Johnsrude (Commun. Sci. and Disord., Western Univ., Rm. 227 Brain and Mind Inst., Natural Sci. Ctr., Western University, London, Ontario N6A 5B7, Canada, ijohnsru@uwo.ca) and Jennifer M. Rodd (Psych. and Lang. Sci., Univ. College London, London, United Kingdom)

Listening effort is an increasingly important concept for applied hearing researchers, but it has not been well elaborated in the cognitive literature. We propose that listening effort is the product of two factors: the processing demands imposed by the listening situation, and the cognitive resources that an individual brings to bear, to compensate for demands. Whereas cognitive resources differ markedly among individuals, processing demands are generally constant in a given listening situation, at least for normal-hearing individuals, and fall into at least three different categories: (i) perceptual demands, (ii) linguistic demands, and (iii) concurrent task demands. Perceptual demands are increased when the speech is degraded, when concurrent interfering sounds are present, or when the listener is hearing impaired. Linguistic demands are increased, for example, when speech is semantically or syntactically complex, or when meaning is ambiguous. Finally, when a listener is performing another task (such as driving) while hearing speech, additional demands are present. A series of behavioral and neuroimaging experiments reveals the nature of these different demands, which appear to be largely independent but to interact in left inferior frontal cortex; fMRI activity in this area may serve as an objective, quantifiable, measure of listening effort. Interactions among different types of demand imply that they cannot be fully understood by studying them in isolation.

11:20

1aSC7. Recruitment of the speech motor system in adverse listening conditions. Claude Alain and Yi Du (Rotman Res. Inst., Baycrest Hospital, 3560 Bathurst St., Toronto, Ontario M6A 2E1, Canada, calain@research.baycrest.org)

Background noise is detrimental to speech comprehension. The decline-compensation hypothesis posits that deficits in sensory processing regions caused by background noise can be counteracted by compensatory recruitment of more general cognitive areas. We are exploring the role of the speech motor system as a compensatory mechanism during impoverished sensory representations in the auditory cortices. Prior studies using functional magnetic resonance imaging (fMRI) have revealed an increase in prefrontal activity when peripheral and central auditory systems cannot effectively process speech sounds. Our research using event-related fMRI revealed a negative correlation between brain activation and perceptual accuracy in speech motor regions (i.e., left ventral premotor cortex (PMv) and Broca's area), suggesting a compensatory recruitment of the motor system during speech perception in noise. Moreover, multi-voxel pattern analysis revealed effective phoneme categorization in the PMv and Broca's area, even in adverse listening conditions. This is in sharp contrast with phoneme discriminability in auditory cortices and left posterior superior temporal gyrus, which showed reliable phoneme classification only when the noise was extremely weak. Better discriminative activity in the speech motor system may compensate for the loss of specificity in the auditory system by forward sensorimotor mapping during adverse listening conditions.

11:40

1aSC8. Electroencephalographic estimates of listening effort for hearing aid fitting. Daniel J. Strauss, Corinna Bernarding (Systems Neurosci. & NeuroTechnol. Unit, Saarland Univ., Saarland University Hospital, NeuroCtr. Bldg. 90.5, SNN-Unit, Homburg/Saar 66421, Germany, daniel.strauss@uni-saarland.de), Ronny Hannemann (Siemens Audiologische Technik, Erlangen, Germany), and Farah I. Corona-Strauss (Systems Neurosci. & NeuroTechnol. Unit, Saarland Univ., Saarbruecken, Germany)

Objective estimates of listening effort could support hearing aid fitting procedures. For this reason, we have analyzed neural correlates of listening effort using electroencephalographic methods and quantitative neurofunctional modeling in recent years. In particular, we have developed a neurophysical corticothalamic feedback model for electroencephalographic listening effort correlates in event-related potentials (ERP), which allowed us to compare simulated and measured data. However, ERP paradigms suffer from their limited flexibility in hearing aid fitting as they are restricted to a certain class of stimulation protocols. More recently, we have shown that attention related listening effort correlates can also be extracted from the oscillatory electroencephalographic activity when using a circular analysis of the instantaneous phase organization of Hardy space projected versions of the neural signals. In this talk, we are going (a) to review these approaches, (b) present results that show that the oscillatory method allows for the extraction of listening effort correlates in real life fitting settings, and (c) discuss the potential of time-resolved listening effort profiles. We think that the latter will stimulate new ideas what objective methods can do and what subjective psychometric methods cannot do.

Session 1aUW

Underwater Acoustics: Acoustic Communications and Scattering

Shane Guan, Cochair
National Marine Fisheries Service, 1315 East-West Highway, SSMC-3, Suite 13700, Silver Spring, MD 20902

Sean Walstead, Cochair
ECE/SIO, UCSD, 9500 Gilman Drive, 0407, La Jolla, CA 92093-0407

Contributed Papers

8:30

1aUW1. Experimental results on underwater communication using vector transducers. Erjian Zhang, Ali Abdi (Elec. Comput. Eng., New Jersey Inst. of Technol., Newark, NJ 07102, ez7@njit.edu), and Chen Chen (Samsung Mobile Solution Lab, San Diego, CA)

Recent studies show that a vector transducer which excites signals in acoustic particle velocity channels can serve as a compact multichannel communication transmitter (C. Chen and A. Abdi, "Signal transmission using underwater acoustic vector transducers," *IEEE Trans. Signal Processing*, **61**, 3683–3698, 2013). In this paper, performance of frequency shift keying modulation is studied, both theoretically and experimentally, for data transmission via underwater particle velocity channels using a vector transducer. Multiple test environments are considered. Comparisons are also made with systems that utilize scalar transducers only. System and transducer design considerations, implementation details, and experimental results are presented in the paper as well. [Work supported in part by the National Science Foundation (NSF), Grants IIP-1340415 and CCF-0830190.]

8:45

1aUW2. Blind deconvolution of simple communication signals recorded in laboratory water tank. Jane Kim, Alex Douglass, Paul Choi, and David R. Dowling (Mech. Eng., Univ. of Michigan, 760 Peninsula Ct., Ann Arbor, MI 48105, janehjki@umich.edu)

Overlapping synthetic time reversal (OSTR) is a technique for estimating the original source signal from acoustic array recordings in an unknown, multipath, time-varying sound channel without using a channel probe or calibration pulse. The OSTR technique applies synthetic time reversal (STR) to short-duration overlapping time segments of the received signal and then stitches the sequences of signal estimates together to blindly recover the original long-duration signal. Previous shallow ocean simulations of OSTR with a 2 kHz center frequency were successful. This presentation describes experimental results for OSTR in the reverberant environment provided by a laboratory water tank with 1.07-m diameter and filled to 0.80-m depth. Here, a variable-duration 50 kHz center frequency communication signal modulated with binary phase shift keying and four carrier cycles per bit was broadcast from an omnidirectional source to either a 1-by-16 or 4×4 receiving array that was 30 to 50 cm away. The nominal multipath reverberation time was varied from 10 ms to 4 ms by adding absorption material to the walls of the tank. Experimental signal reconstruction results are promising and demodulation diagrams are shown for 8, 64, 512, and 4096-bit signals. [Sponsored by the ONR and NAVSEA.]

9:00

1aUW3. Inter-pulse noise field during an Arctic shallow-water seismic survey. Shane Guan (National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20902, shane.guan@noaa.gov), Joseph F. Vignola, John A. Judge, DIego Turo (Dept. of Mech. Eng., Catholic Univ. of America, Washington, DC), and Teresa J. Ryan (Dept. of Eng., East Carolina Univ., Greenville, NC)

Marine seismic surveys using airgun arrays generate intense underwater acoustic pulses. Those pulses may cause hearing impairment and/or behavioral disturbances in marine mammals. Few studies have investigated the resulting multipath propagation and reverberation from these pulses. This research uses acoustic recordings collected in a shallow region of the Beaufort Sea during an open-water seismic survey to characterize the noise field between airgun pulses. The disturbances between pulses are collectively referred to as the inter-pulse sound. Two methods were used to quantify the inter-pulse sound: the incremental computation method and the Hilbert transform method. The incremental computation method calculates the root-mean-squared sound pressure level in various sub-intervals, and the Hilbert transform method is used to calculate instantaneous acoustic amplitudes. Analyses using both methods yield identical results, showing that the inter-pulse sound field exceeds ambient noise levels by as much as 9 dB during relatively quiet conditions. The results also indicate that inter-pulse noise levels are related to the source distance. These two methods can be used to quantify the impact of anthropogenic transient noises on the acoustic environment and to address acoustic masking in marine mammals.

9:15

1aUW4. Multichannel blind deconvolution of sound source of opportunity in ocean waveguide. Sung-Hoon Byun, Karim G. Sabra (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, byunsh@kriso.re.kr), Ning Tian, and Justin Romberg (School of Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA)

Signals that travel through an ocean waveguide are typically distorted when they are received by a remote receiver because of interference and distortion arising from multiple propagation paths. This presentation investigates the applicability of a physics-based blind-deconvolution technique proposed by Sabra *et al.* (JASA **127**, EL42, 2010) to sound sources of opportunity such as randomly radiating ships. Using only the noisy recorded signals from an underwater receiver array, this blind deconvolution provides a means to estimate the Green's functions between the source of opportunity and the elements of the receiver array as well as the original signal radiated by the source of opportunity. These estimated Green's functions can then be used for acoustic characterization of the ocean waveguide parameters which is typically done using controlled sources only. We will discuss the performance of the proposed approach using both numerical simulations and at-sea data using discrete shipping events recorded on a vertical array as sources of opportunities.

9:30

1aUW5. High-frequency broadband seafloor backscatter in a sandy estuarine environment. Eric J. Bajor (Univ. Of New Hampshire, 74 Rolling Ridge Ln., Methuen, Massachusetts 01844, ebajor@ccom.unh.edu), Thomas C. Weber, Larry Ward, Yuri Rzhanov, and Han Hu (Univ. Of New Hampshire, Durham, NH)

Seafloor backscatter collected with high-frequency (>100 kHz) hydrographic echosounders has become an important aspect of seafloor characterization for benthic ecologists and other scientists. The mechanisms that control acoustic scattering at these high frequencies are not completely understood, although surficial roughness and the presence of large scatterers (e.g., shell hash) are likely contributors. To further our understanding of these mechanisms, broadband (100–250 kHz) acoustic measurements were taken at a grazing angle of 45 degrees in a shallow-water, sandy environment with a known presence of shell hash. Stereo imagery was collected simultaneously to quantify the roughness spectrum of the seafloor. Sediment samples were also collected on site of the experiment to quantitatively analyze the content of shell hash. The frequency dependence of the seafloor backscatter will be discussed in terms of its consistency, or lack thereof, with surficial roughness and shell-fragment scattering models.

9:45

1aUW6. Statistical characterization of biologic clutter from the shallow water TREX13 reverberation experiments. John R. Preston (ARL, Pennsylvania State Univ., P. O. Box 30, MS3510, ARL, Penn State Univ., State College, PA 16804, jrp7@arl.psu.edu)

A large experimental effort called TREX13 was conducted in April–May 2013 off Panama City, Florida. As part of this effort, reverberation and clutter measurements were taken in a fixed-fixed configuration in very shallow water (~20 m) over a 22 day period. Results have been presented previously characterizing reverberation, clutter and noise in the 1800–5000 Hz band. The received data are taken from the triplet sub-aperture of the Five Octave Research Array (FORA). The array was fixed 2 m off the sea floor and data were passed to a nearby moored ship (the R/V Sharp). An ITC 2015 source transducer was fixed 1.1 m off the seafloor nearby. Pulses comprised of gated CWs and LFMs were used in this study. Matched filtered plots of the reverberation and clutter along particular bearings are presented using the FORA triplet beamformer. There are clear indications of biologic scattering. Statistical characterization of the biologic components of the reverberation are presented using K-distribution based algorithms to note differences in the estimated shape parameter. Help from the Applied Physics Laboratory at the University of Washington was crucial to this effort. [Work supported by ONR code 322OA.]

10:00

1aUW7. Statistics of very high frequency sound scattered from wind-driven waves. Sean Walstead (ECE, UCSD, 9500 Gilman Dr., 0407, La Jolla, CA 92093-0407, swalstead@ucsd.edu) and Grant B. Deane (SIO, UCSD, La Jolla, CA)

The amplitude, Doppler spread, and temporal coherence of VHF scattering is important for the performance of high frequency sonars and underwater communications systems in operating scenarios where energy from the sea surface cannot be screened. For this talk, the amplitude and arrival time statistics of very high frequency (50 kHz–2000 kHz) sound scattered from a rough wind-driven wave surface are modeled. Saturation occurs in the normalized second moment of acoustic intensity when the surface correlation length exceeds a Fresnel zone length. Fluctuations in arrival time do not saturate and increase proportionally to the dominant surface wave component. Data collected at 300 kHz agree with the model results. Energy is first detected in the wind-driven surface wave field at wavelengths of 1.7 cm and 3.4 cm. The former wave corresponds to a gravity-capillary wave whose phase speed is at a minimum. The principal appearance of this type of wave has threshold implications for the range of acoustic transmission frequencies susceptible to saturation in amplitude statistics.

10:15

1aUW8. The effect of internal waves on the ambient noise vertical directionality in deep ocean. Mehdi Farrokhrooz and Kathleen E. Wage (George Mason Univ., 4450 Rivanna River Way PMB3740, Fairfax, VA 22030, mfarrokh@masonlive.gmu.edu)

Vertical array measurements of low-frequency ambient noise in the mid-latitudes indicate that distant shipping noise has a flat angular distribution concentrated around broadside, e.g., [Wales and Diachok, JASA, 1981]. Ships cannot directly excite the lowest modes (associated with broadside angles) since those modes are trapped near the sound channel axis. Dashen and Munk [JASA, 1984] considered three mechanisms that could transfer shipping noise into the low modes: (1) downslope conversion on the continental slope, (2) volume scattering due to internal waves, and (3) direct excitation of the low modes at high latitudes where the sound channel intersects the surface. Dashen and Munk conclude that the first mechanism is the most likely since the second requires impractically large propagation ranges and the third requires unrealistic shipping densities at high latitudes. A limitation of Dashen and Munk's internal wave analysis is that it does not consider the effect of bottom loss. This talk adapts the transport theory approach of Colosi and Morozov [JASA, 2009] to study the effect of internal waves on the vertical directionality of distant shipping noise in the presence of seafloor attenuation. Initial results indicate that internal waves may have a significant effect at shorter ranges than previously thought.

10:30–10:45 Break

10:45

1aUW9. Nonlinear interaction of monochromatic wave with noise. Desen Yang, Qing Liu, Shengguo Shi, Jie Shi, and Bo Hu (College of Underwater Acoust. Engineefing, Harbin Eng. Univ., No.145 Nantong St., Nangang District, Harbin City, Heilongjiang Province, Harbin, China, liuqing0104@126.com)

The nonlinear interaction of monochromatic wave with noise is studied theoretically, and that is based on the theory of O. V. Rudenko about interactions of intense noise waves and the theory of G. H. Du about nonlinear interaction of finite amplitude irregular waves. The effect on shifting of noise energy by the intensity and frequency of monochromatic wave is discussed during the propagation of the monochromatic wave and noise. Some numerical simulations about this process are conducted under different initial conditions to investigate the changes of their spectrum. The agreement between theoretical analysis and numerical simulation demonstrates that under some conditions the noise can be modulated and the noise spectrum becomes flatter and broader with the increasing distance.

11:00

1aUW10. Two-dimensional modeling of sound scattering with corrective smoothed particle method. Xu Li, Tao Zhang (Dept. of Naval Architecture and Ocean Eng., School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan 430074, China, lixu199123@gmail.com), and YongOu Zhang (Dept. of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei, China)

Meshfree methods are widely used in solving acoustic problems, since a set of arbitrary distributed particles can easily represent systems with moving boundaries or a variety of media. Among all meshfree methods, the smoothed particle hydrodynamics (SPH) method is suitable in handling problems with large ranges of density and object separation. This paper aims at using the corrective smoothed particle method (CSPM) to simulate the two-dimensional underwater sound scattering in the time domain. First, a novel kind of acoustic boundary for the CSPM is built and tested with a plane wave model. Then, the CSPM code is used to solve sound scattering field of an infinite long cylinder, which has different computational parameters for the boundary condition. Finally, the distribution of sound scattering pressure obtained with theoretical solutions is used to validate the CSPM code, and the efficiency of the code is analyzed by comparing with the SPH results in the fields of computation time and accuracy. The suitable value of computational parameters of the boundary is also discussed.

11:15

1aUW11. Design optimization of low density parity check codes in multipath underwater acoustic channels. Shengxing Liu and Qiang Fu (Appl. Ocean Phys. and Eng., Xiamen Univ., Xiping Bldg. C3-223, Xiamen University, Xiamen 361102, China, liusx@xmu.edu.cn)

We propose an iterative decoding and equalizing scheme using low density parity check codes and decision feedback equalizer for underwater acoustic communication system. The scheme decodes and equalizes iteratively by linking a LDPC decoder with an adaptive DFE and feeding output soft information of the LDPC decoder to the DFE as priori information. We extend extrinsic information transfer (EXIT) charts to analyze the performance of LDPC codes in multipath underwater acoustic channels. Furthermore, we introduce an EXIT aided method of optimally designing the parameters of LDPC codes using modified differential evolution algorithm in multipath underwater acoustic channel. Design example and simulation are presented in two different underwater acoustic channels which are generated by BELLOP ray tracing model. Bit error rate of the proposed scheme is less than 10^{-5} as signal to noise rate (SNR) is greater than 8.5 dB in the channels. Compared the regular LDPC code with degree 3 and the Turbo code with feedback and feedforward generator polynomials $1+D+D^2+D^3$ and $1+D^2+D^3$, the optimized LDPC codes achieve gains of about 0.5 dB and 0.8 dB, respectively, at BER= 10^{-4} under the same code rate and block length.

11:30

1aUW12. Experimental results of adaptive multichannel decision feedback equalization in shallow channel. XueLi Sheng, LiNa Fan (Harbin Eng. University, No.145 Nantong St., Nangang District, Harbin, HeiLongjiang 150001, China, shengxueli@aliyun.com), Aijun Song, and Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Newark, DE)

A kind of underwater communication receiver combined with mean-squared error adaptive error adaptive algorithm and multichannel decision feedback equalization (AMDFE) is studied, which can address the problem of underwater communications over a time-varying multipath channel in the presence of worse intersymbol interference and waveform distortion. Its performances in different range and sound profile have been analyzed by simulated shallow water channels in the earlier paper. In this paper, the performances of AMDFE in different underwater channels with different parameters are mainly demonstrated by the experiments on the lake and sea. The anti-multipath ability and parameters selection regulation of AMDFE in both near and long range are proved by real data in shallow water.

11:45

1aUW13. Experimental demonstration of multiband underwater acoustic transmissions based on single carrier communication. Xiao Han, Jingwei Yin, Ge Yu, and Bing Liu (Harbin Eng. Univ., No. 145 Bldg., Nantong St., Nangang District, Harbin, Harbin 150001, China, hanxiao1322@hrbeu.edu.cn)

In a recent shallow water experiment, multiband transmissions were carried out using a vertical array with six sensors uniformly spaced every 0.5 m. The entire frequency band (2–8 kHz) was divided into two sub-bands, each of which is about 2.55 kHz in width. Time reversal processing is first used to compress severe multipath spread, following by a single channel decision feedback equalizer (DFE) to remove residual inter-symbol-interference (ISI). Using experimental data, this paper demonstrates that multiband transmissions are very beneficial to improve the throughput of underwater acoustic communications, achieving a data rate of 6k bits/s using QPSK modulation with almost error free performance.

Posters will be on display and contributors will be at their posters during the session break from 10.30 a.m to 10.45 a.m.

1aUW14. Experimental demonstration of under ice acoustic communication. Jingwei Yin (Harbin Eng. Univ., Rm. 301, Shuisheng Bldg., Harbin 150001, China, yinjingwei@hrbeu.edu.cn), Pengyu Du, Xin Wang, and Jiahao Guo (Harbin Eng. Univ., Harbin, Heilongjiang, China)

Under ice acoustic communication experimental was done in Songhua River, Harbin, China, in January 2015. Minus 20–30 degrees work environment brings a great challenge to the under ice experimental. A series of underwater acoustic communication tests including spread spectrum, OFDM, pattern time delay shift coding (PDS) and CDMA have been achieved. Experimental result shows that the under ice channel is relatively stable and the closer to the ice the simpler the channel structure is. All of the under ice acoustic communication tests achieve low bit error rate communication at 1 km range with different received depth. Under ice CDMA multiuser acoustic communication shows that as many as 12 users can be supported simultaneously with as few as five receivers in under ice channels, using the time reversal mirror combined with the differential correlation detectors.

1aUW15. Fading statistics variation and optimum Mary frequency shift keying system design in sea surface fluctuation. Xue Dandan, Kyu-Chil Park, Jihyun Park, and Jong R. Yoon (Pukyong National Univ., 45 Yongsoro Namgu, Busan 608-737, South Korea, jryoon@pknu.ac.kr)

The underwater acoustic channel can be expressed as a frequency selective fading channel due to multipath, sea surface roughness, and propagation medium change with time and space. Therefore the fading statistics varies with time, space, and frequency. Inherent factor of this fading change is related to a frequency dependent constructive or destructive interference change with time, space, and frequency. The magnitude of interference depends on closely on sea surface roughness. In this study, channel fading statistics are analyzed with delay spread and Doppler spread using pseudo noise (PN) signals. Based on this channel fading statistics, a low data rate M-ary frequency shift keying (MFSK) system of 1kbit/s, is designed for a command/control of navigation or an equipment status monitoring system. The forward error correcting code of convolution code is also applied. Experimental work is conducted in shallow water and it is found that the design parameters of MFSK system can be optimized using the measured fading statistics deduced from PN signal.

1aUW16. Differential pattern time delay shift coding underwater acoustic communication using parametric array. Jingwei Yin, Xiao Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Rm. 801, ShuiSheng Bldg., Harbin, HeiLongJiang 150001, China, zhangxiao@hrbeu.edu.cn), and Yiming Zhou (ShangHai Acoust. Lab., Chinese Acad. Sci., ShangHai, China)

The long range and high data rate communication are two main targets of point to point underwater acoustic communication (PPUAC). In order to achieve the longer communication range, low frequency is the best choice. However, the data rate is limited by narrow available bandwidth at low frequency. So, the system band select must compromise between the data rate and communication range according to the application scenarios. Parametric array can achieve considerable bandwidth at the low frequency band with narrow beam pattern by nonlinear acoustics conversion. These characteristics meet the requirements of both data rate and communication range to a certain extent, and make the channel equalization more simple and effective. A PPUAC scheme based on differential pattern time delay shift coding using parametric array has been proposed. The system employs a 150 kHz primary frequency then get a 10 kHz secondary frequency wave. A proof test has been carried out under ice in SongHua River, HeiLongjiang Province, China. The error free communication was achieved in the trial. The communication scheme is expected to been used in underwater platform directional remote control and other scenarios.

Session 1pAAa

Architectural Acoustics and Noise: Uncertainty in Laboratory Building Acoustics Measurements

Matthew V. Golden, Chair
Pliteq, 616 4th Street, NE, Washington, DC 20002

Chair's Introduction—1:00

Invited Papers

1:05

1pAAa1. Evolution of ASTM E90 for measurement of sound transmission loss and related standards. Noral D. Stewart (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 101, Raleigh, NC 27607, noral14@sacnc.com)

The history ASTM E90 for measurement of sound transmission loss will be traced from the initial release of the first tentative standard in 1950, identifying significant changes that have occurred with each revision. To a lesser degree, the history of ASTM E413 on the sound transmission class, ASTM E1322 on rating outdoor-indoor sound transmission, ASTM E336 for measurement of sound isolation and insulation in the field, and ASTM E966 on outdoor to indoor measurements will be discussed.

1:25

1pAAa2. Understanding the concepts of uncertainty, reproducibility, and repeatability and the application to acoustic testing. Ralph T. Muehleisen (Energy Systems, Argonne National Lab., 9700 S. Cass Ave., Bldg. 202, Lemont, IL 60439, rmuehleisen@anl.gov)

The consideration of uncertainty in measurement has long been an important part of scientific measurement. As measurements became standardized and testing and measurement standards were developed, various forms of uncertainty were sometimes, but not always, considered. Understanding various forms of uncertainty and the basic concepts of uncertainty, reproducibility, and repeatability are important for testing even if the estimation of them is not directly part of the test standard. In this presentation, the concepts of uncertainty, reproducibility, and repeatability for acoustic testing are explained using the ASTM E90 test as an example for explanation.

1:45

1pAAa3. A review of academic research and data related to laboratory testing for sound transmission loss. Benjamin Shafer (Tech. Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406, ben.shafer@quietrock.com)

The building noise control design and manufacturing industry currently relies heavily on laboratory-tested sound transmission loss data for the design and implementation of noise control treatments in buildings. Although most sound transmission loss laboratories achieve acceptable measurement repeatability, the reproducibility of this test data has recently fallen under scrutiny and is currently unknown for common building noise control partition designs. Key academic articles will be discussed that provide a fundamental understanding of how laboratory sound transmission loss measurement procedures and laboratory facilities may affect reproducibility. A review of comparative sound transmission loss data for various laboratories will also be provided in an effort to clearly illustrate why sound transmission loss testing reproducibility has come under scrutiny. It is essential that organizations for standards and design professionals become more aware of these realities.

2:05

1pAAa4. The influence of test fixture damping on the measurement of sound transmission loss. Jennifer A. Shaw, Charles T. Moritz, and Armando Carrera (Blachford, Inc., Blachford Acoust. Lab., 1445 Powis Rd., West Chicago, IL 60185, jshaw@blachfordinc.com)

The sound transmission loss (STL) provided by a material depends on factors such as its mass, damping, and stiffness. In the lab, the damping and stiffness of a panel is influenced by the test fixture. To reduce test to test and lab to lab variation, the test standards give some detail with regards to installing the test specimen and sealing around the edges to prevent leakage between test rooms. However, many times it is not possible to simulate typical installation and sealing methods due to the wide range of installation options for the product, and a heavy layer of clay or other material is used to seal the sample in the fixture. As the size of the sample decreases, the edge damping from clamping and/or sealing the sample can add significant damping to a panel. Too much damping can cause a significant overstatement of the sample STL. This paper examines the influence of edge damping due to sealing clay on the STL of steel and aluminum panels.

2:25

1pAAa5. Categories of repeatability and reproducibility in acoustical laboratories. John J. LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

There are many categories of variability in acoustical laboratory testing. The repeatability of acoustical testing is generally defined based on repeated testing of the same assembly in the same laboratory and conditions over a short time period, while the reproducibility is based on testing a (nominally identical) assembly in a different laboratory. However there is a spectrum of uncertainties between these points. The authors previously documented "long term repeatability," the repeated testing of the identical specimen in the same laboratory with the same equipment and operator, but over an extended time period [J. Acoust. Soc. Am. **130**, 2355 (2011)]. The current study examines the "long term rebuild repeatability," a nominally identical assembly rebuilt in the same laboratory over a long time period. The results are compared with previously measured variability, and the implications for manufacturers and designers are discussed.

2:45–3:00 Break

3:00

1pAAa6. The National Voluntary Laboratory Accreditation Program and Acoustical Testing Services Laboratory Accreditation Program. Kari K. Harper (NIST NVLAP, NIST, 100 Bureau Dr., M/S 2140, Gaithersburg, MD 20899, kharper@nist.gov)

This presentation provides an overview of NVLAP, the National Voluntary Laboratory Accreditation Program of the National Institute of Standards and Technology (NIST), with particular focus on NVLAP's Acoustical Testing Services Laboratory Accreditation Program (LAP). Through the Acoustical Testing Services LAP, NVLAP provides third-party attestation of the conformance of its accredited acoustics laboratories to the requirements of ISO/IEC 17025:2005, *General requirements for the competence of testing and calibration laboratories*; NIST Handbook 150, *NVLAP Procedures and General Requirements*; and NIST Handbook 150-8, *NVLAP Acoustical Testing Services*. Specific topics covered in this presentation include a brief history of the Acoustical Testing Services LAP, the accreditation process, proficiency testing, and benefits of accreditation, such as international recognition of the competence of accredited laboratories and the concept of "tested once; accepted everywhere."

3:20

1pAAa7. What acousticians need to know about the third party sound transmission class testing process and acoustical materials performance validation. Eric P. Wolfram (Riverbank Acoust. Labs., 1512 S Batavia Ave., Geneva, IL 60134, ewolfram@alionscience.com)

The architectural product industry demonstrates sound transmission loss performance by commissioning third party tests from accredited laboratories performed according to the ASTM E90 standard. This presentation will provide insight into the materials testing process, laboratory quality management, and explanation of the report documents. The presenter will share experience and advice for the use of test results to evaluate acoustical product performance. Acousticians, architects, and engineers must play a role in improving quality of published data by demanding more complete information from manufacturer's marketing departments regarding product acoustical performance.

3:40

1pAAa8. Status report on the planned intra-laboratory study to improve American Society for Testing and Materials E90's precision and bias statement. Matthew V. Golden (Pliteq, 616 4th St., NE, Washington, DC 20002, mgolden@pliteq.com)

The ASTM E33 Committee on Building and Environmental Acoustics has recently undertaken an effort to improve the precision and bias statements of all of the standards under its purview. Based on an analysis of the current state of the precision and bias statements and each standards importance to the industry, ASTM E90 was chosen as the standard to focus on first. Improving the precision and bias statements requires an intra-laboratory performance study, also known as a round robin, to be performed. Previous round robins for ASTM E 90 used relatively low performing specimens, Sound Transmission Class (STC) 24-32. The current proposed round robin is planned to use specimens with a STC in the high 40's to low 50's. Details of these efforts will be shared.

Contributed Paper

4:00

1pAAa9. Reproducibility and repeatability of measuring noise level reduction using an artificial noise source. Rene Robert, Kenneth Cunefare (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Office 002, Atlanta, GA 30332, rrobert6@gatech.edu), Erica Ryherd (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE), and Javier Irizarry (School of Bldg. Construction, Georgia Inst. of Technol., Atlanta, GA)

Buildings that are subjected to aviation noise require extra sound isolation measures in order to keep the indoor noise levels below a certain threshold. The difference in sound pressure from outside to inside a building is often quantified as a single number known as the noise level reduction (NLR). Generally, the procedures described in ASTM E966-10 are followed in determining this number for a façade, which is then used to determine the modifications required. The standard allows testing with one of two noise sources: an actual traffic source or a loudspeaker. An investigation is underway to statistically evaluate the repeatability and reproducibility of measurements taken on a façade for a "test house" with a loud speaker. The test house is a single-room structure that was designed and constructed with materials and methods of a residence in a mixed-humid climate. The repeatability is defined as the ability for a specific test to be implemented multiple times with comparable results, while the reproducibility is the ability for various test configurations allowed within the standard to yield comparable results. The results of this analysis can be used to improve testing parameters, further validate the procedure, and provide an estimate of confidence bounds.

4:15–4:45 Panel Discussion

Session 1pAAb

Architectural Acoustics, Noise, and Structural Acoustics and Vibration: Session in Honor of Laymon Miller

Neil T. Shade, Cochair
Acoustical Design Collaborative, Ltd, 7509 Lhirondelle Club Road, Ruxton, MD 21204

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square - Suite 103, Vernon, CT 06066

Chair's Introduction—2:00

Invited Papers

2:05

1pAAb1. Family and life experiences with Laymon. Robert L. Miller (Harris Miller Miller & Hanson Inc., 77 South Bedford St., Burlington, MA 01803, rmiller@hmmh.com)

This talk will bring to light memorable personal recollections growing up with Laymon as husband, father, and mentor. My own memories will illustrate how his love of acoustics, the people he worked with, and the problems he worked on ultimately led to my own career in acoustics. They will include a recounting of childhood business trips, flying experiences, and a summer job opportunity at Bolt Beranek and Newman. The talk also will include his wife, Lucy's, still-vivid recollections of Laymon's first years at the Harvard Underwater Sound Lab, his nine years at Penn State, and many years later their times together as she accompanied him on his BBN lecture series. Unpublished excerpts from Laymon's five-volume autobiography will be used to illustrate his own detailed and insightful accounting of other family and life experiences, including his own "ethical will," written for later generations to better understand and appreciate the values and beliefs for which he most wanted to be remembered.

2:25

1pAAb2. Laymon Miller, distinguished consultant in noise control engineering at Bolt Beranek and Newman. Leo Beranek (Consultant in Acoust., 10 Longwood Dr., Westwood, MA 02090, beranekleo@ieee.org)

Laymon Miller was employed in Harvard's Underwater Sound Lab (HUSL), under the direction of Professor F. V. Hunt during World War ll. When that laboratory closed, Laymon joined the consulting staff at Bolt Beranek and Newman (BBN). Laymon became BBN's leading expert on the reduction of noise and vibration in ventilating systems. One of his important jobs was noise reduction at New York's Lincoln Center's Philharmonic Hall. He also became expert in vibrations transmitted through the earth. In particular, he designed the vibration isolation pads used to reduce the transmission of railroad track vibrations into adjacent buildings in a number of cities, including Boston's Back Bay and Montreal. Practically every important building acoustics project that came into BBN received his attention. He gained the reputation of more chargeable consulting time each month than any other employee. Besides mentoring numerous new BBN employees he taught highly sought after courses on noise reduction for architects, building, industrial and plant engineers, as well as for other acoustical consultants.

2:45

1pAAb3. Reminiscences on Laymon Miller's remarkable twenty seven year consulting career at Bolt Beranek and Newman in Cambridge. William J. Cavanaugh (Cavanaugh Tocci Assoc. Inc., 3 Merifield Ln., Natick, MA 01760-5520, wcavanaugh@cavtocci.com)

Laymon Miller joined the noise control engineering consulting staff of Bolt Beranek and Newman (BBN) on August 1, 1954, bringing an impressive record of experience in acoustics from Harvard University's top secret Underwater Sound Lab (HUSL) during WWll followed by ten years as head of the acoustics section at Penn State's Ordinance Research Lab (ORL). Laymon provided invaluable mentoring for many new BBN consulting staff members like this author whose only experience in acoustics were introductory courses in architectural acoustics taught by Physics Professor Richard Bolt at MIT's School of Architecture and the MIT Acoustics Lab, Laymon quickly established a reputation among BBN employees and clients alike as an outstanding teacher and contributed to BBN's unofficial title as the "third important graduate school in Cambridge." He documented his diverse consulting career through numerous technical papers and articles, through technical brochures he prepared for clients like the US Army Corps of Engineers,manufactures like the Baltimore Air Coil Company and through reports he prepared for thousands of BBN projects on which he served as principal consultant. Laymon retired from BBN in 1981 after a remarkable 27 year career making quieter buildings, concert halls, workplaces, communities, transportation vehicles, and in general, "the world a better place."

3:05

1pAAb4. Laymon N. Miller—Contributions to industrial and community noise control. Eric W. Wood (Acentech, 33 Moulton St., Cambridge, MA 02138, ewood@acentech.com)

Laymon N. Miller, following graduation in 1939 from the University of Texas in Austin, spent time first at the Underwater Sound Lab at Harvard University and then at Penn State as Head of the Acoustics Section of Ordnance Research Lab (ORL). In 1954, he joined Bolt Beranek and Newman where he was honored as their first Principal Consultant. This presentation describes his contributions to industrial and community noise control while consulting for a wide range of many clients for 27 years, until his retirement in 1981. Laymon Miller, a gentleman, friend, and colleague.

3:25

1pAAb5. Origin and history of the Laymon Miller noise course. Reginald H. Keith (Hoover & Keith Inc., 11381 Meadowglen Ln., Ste.. I, Houston, TX 77082, reggie.keith@hoover-keith.com)

For 20 years, starting in 1969, Laymon Miller produced and taught a seminal course in noise and vibration control in many different venues and settings. In this paper, I will present information as to the origins of this course, my recollections of first attending the course in 1977, and our experiences in the continuation of the course since 1989.

3:45

1pAAb6. Laymon Miller—An exemplary acoustical consultant. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooksacoustics.com)

Laymon Miller was a "Consultant's consultant." He embodied a wonderful example of how to conduct oneself in the engineering consulting business, providing leadership in defining the functions and responsibilities of acoustical consultants. As a leader of our profession, Laymon was also a great friend to the National Council of Acoustical Consultants (NCAC). Former NCAC Newsletter Editor Bill Cavanaugh asked Laymon if he would serve as a guest editor for a continuing series tentatively titled "War stories … from the Consulting Veteran's files." Laymon answered the call, and this series continued for many years, capturing the "priceless gems" from which all of us, at all experience levels, can learn about the problems faced by consultants, and importantly about the solutions as applied in the field. Laymon was a talented and generous teacher, in print and in person, of those inside and outside of acoustical consulting. His experiences, related with insight and humor, have provided guidance to generations of those in general industry and to acoustical consulting practitioners alike. Laymon was elected an Honorary Member of NCAC in 1993, received the NCAC C. Paul Boner Award in 2007 and was presented the Institute of Noise Control Engineering (INCE) Outstanding Educator Award in 2008.

4:05

1pAAb7. Experiences as editor of Laymon Miller's book *An NCAC Anthology in Noise and Vibration*. Neil T. Shade (Acoust. Design Collaborative, Ltd., 7509 LHirondelle Club Rd., Ruxton, MD 21204, nts@akustx.com)

This presentation will recount the experiences of serving as editor of Laymon Miller's book, *An NCAC Anthology in Noise and Vibration*, published in 2013 by the National Council of Acoustical Consultant's (NCAC). Ever active in retirement, and at the behest of William Cavanaugh, Laymon contributed 60 papers from 1996 to 2012 to the NCAC *Newsletter*. These papers, along with his industry publications from 1957 to 2008, were the basis for his book. A committee was formed within NCAC to oversee the compilation and production of his book with this author serving as editor. The "engineer" in Laymon was evident in the meticulous organization of his writings and instructions provided to the editor. *An NCAC Anthology in Noise and Vibration* is organized in two parts. The first contains industry publications, arranged by topic. The second is Laymon's NCAC articles arranged chronologically. The book concludes with his autobiography, which he humbly described "as a life of surprises." Laymon dedicated his book to his friend and colleague Leo Beranek. Through editing the book the author gained insight into Laymon's seminal contributions to building acoustics, noise and vibration control, community noise, his acoustic consulting adventures, and the work ethic of this remarkable man.

Session 1pABa

Animal Bioacoustics: Physiology of Animal Hearing

Edward J. Walsh, Chair
Research, Boys Town Natl. Res. Hospital, 555 North 30th Street, Omaha, NE 68131

Contributed Papers

1:00

1pABa1. Discrimination of frequency-modulated sweeps by laboratory mice. Laurel A. Screven and Micheal L. Dent (Univ. at Buffalo, B29 Park Hall, Buffalo, NY 14260, laurelsc@buffalo.edu)

Mice often produce ultrasonic vocalizations (USVs) that sweep upwards in frequency from around 60 kHz to around 80 kHz, and similarly sweep downwards in frequency from 80 kHz to 60 kHz. Whether or not these USVs are used for communication purposes is still unknown. Determining the ability of mice to discriminate between synthetic upsweep and downsweep frequency-modulated stimuli will expand the current knowledge about acoustic communication in mice. Mice were trained and tested using operant conditioning procedures and positive reinforcement to discriminate between upsweeps and downsweeps. The stimuli varied in bandwidth, duration, and direction of the sweep. The animals responded when they heard a change in the repeating background, indicating that they could discriminate background from target. The mice performed significantly worse discriminating between background and targets when the stimuli occupied the same bandwidths. Further, the mice's discrimination performance became much worse when the duration approached that of their natural vocalizations. When the sweeps occupied different frequency ranges and longer durations, discrimination performance improved. These results collected using artificial stimuli created to mimic natural USVs indicate that the bandwidth of vocalizations may be much more important for communication than the frequency contours of the vocalizations. [Work supported by NIH DC012302.]

1:15

1pABa2. Hearing sensitivity in the Greater Prairie Chicken (*Tympanuchus cupido*). Edward J. Walsh (Developmental Auditory Physiol. Lab, Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, edward.walsh@boystown.org), Cara Whalen, Larkin Powell, Mary B. Brown (School of Natural Resources, Univ. of Nebraska, Lincoln, NE), and JoAnn McGee (Developmental Auditory Physiol. Lab, Boys Town National Res. Hospital, Omaha, NE)

As their scientific name implies, Greater Prairie Chickens (*Tympanuchus cupido*) are well known for their vocalizations, particularly those produced by males during courtship. In this report, the auditory sensitivity of a small cohort of Greater Prairie Chickens inhabiting the grasslands of southeastern Nebraska is presented as part of an effort to assess the capacity of wind turbine farm generated noise to mask male courtship vocalizations. Birds were captured from breeding grounds (leks) between March and June of 2014 and sensitivity to tone-burst stimuli was assessed using the auditory brainstem response. While response waveforms were typical of responses observed in other avian species, as was the bandwidth of audibility curves, preliminary data suggest the possibility that Greater Prairie Chickens may be unusually sensitive to low frequency tone-bursts in the vicinity of the dominant spectral component of the "booming" call, a prominent, intense vocalization that males produce during courtship. As part of the larger study that is designed to evaluate the potential impact of wind turbine farm operation on reproductive success, demography and overall survival rates, the potential masking influence of noise produced by turbine operations at wind farms will be discussed in relation to sensitivity findings.

1:30

1pABa3. Place specificity of dolphin auditory evoked potentials assessed with high-pass masking noise. James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Jason Mulsow, Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA), and Robert F. Burkard (Univ. of Buffalo, Amherst, New York)

Auditory evoked potential measurements are commonly used for hearing assessment in marine mammals for which psychophysical testing is not practical. Evoked potential measurements typically employ electrical signals consisting of short-duration tone-pips or sinusoidal amplitude modulated tones, which are then presented to piezoelectric underwater sound projectors. Although tone-pip and short-duration tonal stimuli may possess relatively narrow frequency bandwidth, the resulting physiological responses may possess much greater bandwidth, especially for lower frequency stimuli at higher stimulus levels. In this study, high-pass masking noise techniques were used to examine the place specificity of auditory evoked responses from click, tone-pip, and sinusoidal amplitude modulated tones in bottlenose dolphins (*Tursiops truncatus*). The experimental methods for generating and spectrally equalizing masking noise and click stimuli will be presented, along with the effect of compensated clicks with uncompensated clicks and ABR latencies and amplitudes. [Funded by U.S. Navy Living Marine Resources Program.]

1:45

1pABa4. Equal-latency curves and auditory weighting functions for bottlenose dolphins (*Tursiops truncatus*) and California sea lions (*Zalophus californianus*). Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, jason.mulsow@nmmf.org), James J. Finneran (U.S. Navy Marine Mammal Program, SSC Pacific, Code 71510, San Diego, CA), and Carolyn E. Schlundt (ITT Exelis Corp., San Diego, CA)

Reaction time (RT) data obtained from simple tonal detection tasks have been used to estimate frequency-specific equal-loudness contours in non-human animals. In order to guide the design of auditory weighting functions for marine mammals, equal-latency contours were generated using RT data from a simple tonal detection including two bottlenose dolphins (under water) and three California sea lions (in air). Median RT increased exponentially with decreased SPL in all cases. Equal-latency contours for near-threshold RTs were similar to audiograms in both species. Data for the sea lions showed some compression of equal-latency contours with increases in SPL; however, large inter-subject differences in the data for dolphins made results for that species more difficult to interpret. The equal-latency contours for all subjects progressively diverged from predicted equal-loudness contours at higher SPLs, likely a result of very small changes in RT with relatively large increases in SPL. As a result, the contours of most interest for designing weighting functions for high-level noise exposures were also the least reliable. The general similarity of most of the contours to species-typical audiograms suggests that more easily obtained auditory thresholds may provide useful approximations for weighting. [Funded by U.S. Navy Living Marine Resources Program.]

Session 1pABb

Animal Bioacoustics: Advances in Processing Bioacoustic Data

Kristin B. Hodge, Chair
Bioacoustics Research Program, Cornell University, 159 Sapsucker Woods Road, Ithaca, NY 14850

Contributed Papers

2:45

1pABb1. Continental scale acoustic monitoring program: One year of data. Samuel L. Denes (Biology, Syracuse Univ., 116 Appl. Sci. Bldg., University Park, PA 16802, sld980@psu.edu), Susan E. Parks, Leanna Matthews, Hannah Blair (Biology, Syracuse Univ., Syracuse, NY), Pramod Varshney (Elec. Eng. & Comput. Sci., Syracuse Univ., Syracuse, NY), and Kurt Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

A multiyear project is underway to demonstrate the benefits of incorporating acoustic monitoring into the National Ecological Observatory Network (NEON). The NEON project seeks to generate data for the study of continental scale phenomena. Acoustic recordings can be used to determine the presence of acoustically active biota without the presence of field technicians. We have deployed stand-alone acoustic recorders at four NEON sites. Data from these recorders can be used to document spatio-temporal shifts in the presence of acoustically active species of birds, anurans, and insects. Amplitude, frequency band energy, and statistical detection methods have been compared to demonstrate the implementation of automated detection algorithms. Results from acoustic biota surveys are compared with traditional biota surveys. Comparisons of the relative contributions of acoustic energy from geophysical, biotic, and anthropogenic sources within and between sites will be examined. Preliminary results from the first year and a half of acoustic data collection will be presented. [Project supported by NSF award #1340669.]

3:00

1pABb2. Improving knowledge and understanding of underwater sound from exploration and production activities and potential interaction with marine life: An overview of a research program. Ian Voparil (IOGP E&P Sound & Marine Life Joint Industry Programme (JIP), P.O. Box 61933, Rm. 3412, New Orleans, LA 70161, ian.voparil@shell.com)

The International Oil and Gas Producers E&P Sound & Marine Life Joint Industry Programme (JIP), a partnership of 13 oil and gas producing companies, is the world's largest non-governmental funder of research to increase understanding of the potential effects of E&P sound on marine life. The JIP provides objective, scientific information that: Informs and updates policy decision makers and regulatory development processes that affect E&P operations globally Determines the basis for mitigation measures that are protective of marine life, cost effective, and credible with outside stakeholders feeds into planning for efficient E&P project development that is environmentally protective. Since 2006, it has spent $25 million on research and in 2013, the program embarked on a new phase with an additional budget of $18 million. The JIP regularly consults with regulators regarding their needs for data, to enable development of fact-based regulatory decisions. The program co-funds selected projects with government agencies. Research is conducted by independent scientists who are encouraged to publish their results in peer-reviewed literature. To date, JIP-funded research has been published in 75 peer-reviewed papers and all final project reports publically available. During this talk, results of the previous JIP-funded research will be presented. Furthermore, an overview of ongoing and future studies is provided.

3:15

1pABb3. Acoustic monitoring of Bryde's whales (*Balaenoptera edeni*) in the northern Gulf of Mexico. Kristin B. Hodge, Jamey T. Tielens, and Aaron N. Rice (BioAcoust. Res. Program, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, kbhodge@cornell.edu)

Bryde's whales (*Balaenoptera edeni*) are the most common mysticete species occurring in the Gulf of Mexico, yet due to a dearth of visual sightings and stranding data, little is understood regarding their spatial distribution and seasonal occurrence patterns. Two putative call types unique to this population of Bryde's whales have recently been identified, providing an opportunity to understand Bryde's whale spatial and temporal occurrence patterns using passive acoustic monitoring methods. In order to evaluate Bryde's whale presence in the area, marine autonomous recording units (MARUs) were deployed from June 2010 through September 2011 in the northern Gulf of Mexico where Bryde's whales have been historically observed. Distribution, seasonality, and diel patterns were examined separately for each of the call types at all sites and days during which acoustic data were collected by the MARUs. Preliminary results show Bryde's whales were acoustically detected throughout the recording period. Of the two call types associated with Bryde's whales, one call type was produced consistently more often than the other; however, both call types showed a similar diel pattern. These results suggest Bryde's whales may reside or visit this area year-round, which may facilitate future monitoring efforts for this poorly studied population.

3:30

1pABb4. Automatic manatee count using passive acoustics. Jorge M. Castro, Arturo Camacho, and Mario Rivera (CITIC, Universidad de Costa Rica, Ciudad Universitaria Rodrigo Facio, San Pedro, San José 11501, Costa Rica, pctreepkfloyd@gmail.com)

The West Indian manatee is a threatened species throughout its known range. To improve its conservation, it is necessary to locate and count the individuals. This is a very difficult task due to the cryptic behavioral characteristics of the species and the environmental constrains, particularly in the Central American region, where muddy waters produce limited visibility. A method to estimate manatee population through vocalizations is a reliable, inexpensive, non-invasive, and novel option in the region. Digital signal processing techniques are proposed to identify individuals in field recordings of manatee vocalizations. The proposed methods for signal denoising and vocalization recognition have produced promising results. Further work needs to be done to identify individuals.

3:45–4:00 Break

4:00

1pABb5. Localization of individual call among croakers' chorus using a stereo recording system. Masanori Ito, Ikuo Matsuo (Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, matsuo@cs.tohoku-gakuin.ac.jp), Tomohito Imaizumi, and Tomonari Akamatsu (National Res. Inst. of Fisheries Eng., Fisheries Res. Agency, Kamisu, Japan)

Passive acoustic monitoring (PAM) is widely used in cetacean census in the ocean. In these years, PAM has been applied for various species out of mammals such as fish and crustaceans. For detection, classification and density estimation in PAM, isolation of each biological sound and reliable detection is essentially needed. Choruses of fish sounds were recorded by a stereo sound monitoring system on the seafloor in off Choshi, Chiba prefecture, Japan in May 2013. Numbers of croaker sounds were recorded and the simple separation in time domain was not possible in chorus situation. A blind source separation method based on independent component analysis was applied to separate mixed sounds in order to obtain individual sounds and to localize them. The directions of sound arrivals could be estimated by using the cross correlation function of the separated signals and the estimated directions were plotted. Using proposed methods, individual call sounds could be localized and inter-pulse intervals could be accurately estimated. Using separated sequence of sounds from an individual, the movement of phonating fish could be measured using PAM. [This study was supported by CREST, JST.]

4:15

1pABb6. Newtonian and weighted essentially non-oscillatory models to describe the generation and propagation of the Cicada mating calls. Derke Hughes, Sheri L. Martinelli (NUWCDIVNPT, 1176 Howell St., Newport, RI 02841, derke.hughes@verizon.net), Allan D. Pierce (Mech. Eng., Boston Univ., East Sandwich, MA), Richard A. Katz, and Robert M. Koch (NUWCDIVNPT, Newport, RI)

Experiments and analyses of Hughes *et al.*, JASA, 2009 are the origins of this research where we study the in-air waveform generation and propagation of the acoustic signals generated by cicadas. The sound generation is studied in a Newtonian model and the sound propagation is analysis by a numerical solver for viscous Burgers' equation. The time histories from the tymbal surface velocities recorded by a laser Doppler vibrometer to the microphones positioned near the cicadas provide the test data. The Newtonian model describes the sound production systems process to generate the mating call signal structure. The numerical solver employs weighted essentially non-oscillatory (WENO) reconstruction to approximate the first and second derivatives of the semi-discrete operator. The WENO is utilizes due to the non-smooth structure of the cicada propagating waveform. Principally, the cicada mating signal in question has sharp transitions, since spectral methods tend to produce spurious oscillations as a result of attempting to represent a discontinuous function by a Fourier basis expansion. Thus, these analytical models are computationally tested to determine if the results capture the sound production and the transmission of the cicada mating calls. To verify the models are meaningful, the simulations are verified with real experimental data.

4:30

1pABb7. Simple mechanical model reproduces complex calls in a fish vocal system: Implications for the evolution of vertebrate acoustic communication systems. Aaron N. Rice (BioAcoust. Res. Program, Cornell Lab of Ornithology, Ithaca, NY), Sang Min Han, Bruce R. Land (School of Elec. and Comput. Eng., Cornell Univ., 8452 Low Rise 8, Ithaca, NY 14853, sh833@cornell.edu), and Andrew H. Bass (Dept. of Neurobiology and Behavior, Cornell Univ., Ithaca, NY)

A mathematical model has been developed for vocalizations of the three-spined toadfish (*Batrachomoeus trispinosus*) and the plainfin midshipman (*Porichthys notatus*); acoustically active fish species that have served as model systems for studying acoustic communication mechanisms among vertebrates. The toadfish has a vocal organ comprised of a pair of bilaterally separated swim bladders that generate acoustic nonlinearities similar to those found in tetrapods. Midshipman have one swim bladder with sounds lacking the nonlinearities. We modeled the toadfish mechanism as a system of two harmonic oscillators coupled with a tendon-like string, and the midshipman one by the same system without the coupling. The coupling in the toadfish model, along with nonlinear oscillator position and velocity dependent terms, generates signatures of nonlinearity such as deterministic chaos and bifurcating harmonics. The system generates sounds that simulate naturalistic calls of both toadfish and midshipman depending on the input parameters. We built an optimizer to minimize the difference between the power density spectra of the simulated call and an empirical recording. We successfully computed optimal toadfish and midshipman calls and simulated transection experiments, together leading to a deeper understanding of the diversity of vertebrate vocalization mechanisms. [Research support from Rawlings Cornell Presidential Scholars and NSF IOS 1120925.]

Session 1pPA

Physical Acoustics: Acoustofluidics: Interaction of Acoustics and Fluid Dynamic Phenomena

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

Max Denis, Cochair
University of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854

Chair's Introduction—1:00

Invited Papers

1:05

1pPA1. Acoustic tweezers: Manipulating particles, cells, and fluids using sound waves. Tony Jun Huang (Eng. Sci. and Mech., Penn state Univ., Millennium Sci. Complex, PSU, N-330, University Park, PA 16802, junhuang@engr.psu.edu)

The ability to manipulate cells, micro/nanoparticles, and fluids in a biocompatible and dexterous manner is critical for numerous biological studies and applications such as cell-cell communication, biosensing, tissue engineering, regenerative medicine, and lab on a chip. Here, we summarize our recent progress on an "acoustic tweezers" technique that utilizes acoustic waves to manipulate particles, cells, organisms, and fluids. This technique is capable of manipulating cells and microparticles regardless of shape, size, charge, or polarity. Its power intensity, approximately 10^7 times lower than that of optical tweezers, compares favorably with those of other active patterning methods. Cell viability, proliferation, and gene expression have revealed the technique to be extremely biocompatible. The aforementioned advantages, along with this technique's simple design and low-cost, compact design, render the "acoustic tweezers" technique a promising tool for various applications in biology, chemistry, engineering, biophysics, and materials science.

1:30

1pPA2. Intense cavitation in microfluidics for bio-technology applications. Siew-Wan Ohl, Tandiono Tandiono, Evert Klaseboer (Inst. of High Performance Computing, 1 Fusionopolis Way, #16-16 Connexis North, Singapore 138632, Singapore, ohlsw@ihpc.a-star.edu.sg), Dave Ow, Andre Choo (Bioprocessing Technol. Inst., Singapore, Singapore), and Claus-Dieter Ohl (School of Physical and Mathematical Sci., Nanyang Technolog. Univ., Singapore, Singapore)

This study reports the use of intense ultrasonic cavitation in the confinement of a microfluidics channel [1], and the applications that has been developed for the past 4 years [2]–[5]. The cavitation bubbles are created at the gas-water interface due to strong capillary waves which are generated when the system is driven at its natural frequency (around 100 kHz) [1]. These bubbles oscillate and collapse within the channel. The bubbles are useful for sonochemistry and the generation of sonoluminescence [2]. When we add bacteria (*Escherichia coli*), and yeasts (*Pichia pastoris*) into the microfluidics channels, the oscillating and collapsing bubbles stretch and lyse these cells [3]. In another application, human red blood cells are added to a microchamber. Cell stretching and rapture are observed when a laser generated cavitation bubble expands and collapses next to the cell [4]. A numerical model of a liquid pocket surrounded by a membrane with surface tension which was placed next to an oscillating bubble was developed using the Boundary Element Method. Lastly, new results on gene transfection [5], lysing of bacterial spores, and emulsification by ultrasonic bubbles will be presented. References: [1] Tandiono *et al*., Lab Chip **10**(14), 1848–1855 (2010). [2] Tandiono *et al*., Proc. Natl. Acad. Sci. U.S.A. **108**(15), 5996–5998 (2011). [3] Tandiono *et al*., Lab Chip **12**(4), 780–786 (2012). [4] Tandiono *et al*., Soft Matter **9**(36), 8687–8696 (2013). [5] Ling *et al*., Biotechnol. J. **9**(8), 1081–1087 (2014).

1:55

1pPA3. Steady streaming around a pulsating bubble located at the velocity node of a standing wave. Mohammad AlHamli and Satwindar S. Sadhal (Aerosp. & Mech. Eng., Univ. of Southern California, Olin Hall OHE 430, Los Angeles, CA 90089-1453, alhamli@usc.edu)

We have examined the effect of the no-slip boundary condition on the steady streaming around a radially pulsating bubble, located at the velocity node of a standing sound wave. The no-slip condition can take place on the bubble surface under many circumstances, especially if the surface is contaminated or otherwise coated. We applied the singular perturbation analysis with the oscillation amplitude, $\varepsilon=U_0/(a\omega)<<1$, as a small parameter and took the frequency parameter, $M^2=i\omega a^2/\nu$, to be large. Here, a, U_0, ω and ν are length scale, velocity scale, frequency, and kinematic viscosity, respectively. We further assumed that the bubble stays spherical in shape, the wavelength is significantly larger than the bubble radius, and no fluid transport takes place from the bubble to the surroundings. Additionally, the lateral and the radial oscillations, while at the same frequency, have a phase difference φ, and the amplitudes of both the

oscillations are small compared to the bubble radius. We found that the streaming happened at a lower order than the non-pulsating case and it is more intense than the case of the shear free boundary condition. Furthermore, the phase difference was found to play a significant role in the streaming flow direction and its intensity.

2:20

1pPA4. Surface acoustic wave microfluidics: Thin films and drop splitting. James Friend (Mech. and Aerosp. Eng., Univ. of California, San Diego, 345F Structural and Mech. Eng., M.S. 411 Gilman Dr, La Jolla, CA 92093, jfriend@eng.ucsd.edu)

A renaissance in surface acoustic waves (SAW) has occurred, due mainly to their enormously powerful ability to manipulate fluids and colloids for microfluidics. Beyond the routine manipulation of drops, mixing, and separation and concentration of colloids using SAW, we have found fascinating behavior in the formation and propulsion of thin fluid films from a few tens to less than a micrometer thick in various directions, either with or against the acoustic wave propagation. Furthermore, we have seen—and been able to control through careful input conditions—the splitting of fluid drops. Both have a broad range of applications that will be briefly discussed, but more important is the underlying physics that illustrates the rich interaction of the acoustics, fluid dynamics, and free fluid interface in these systems.

2:45

1pPA5. Transition to turbulence in acoustically driven flows. Vineet Mehta (MIT Lincoln Lab., 244 Wood St., Lincoln, MA 02420-9108, vineet@ll.mit.edu), Charles Thompson, Kavitha Chandra (Univ. of Massachusetts Lowell, Lowell, MA), and Max Denis (Mayo Clinic, Rochester, MN)

Direct numerical solutions of the three-dimensional time-dependent Navier-Stokes equation are presented for the evolution of three-dimensional finite-amplitude disturbances in the Stokes boundary layer. The basic flow is driven harmonically in time and the disturbances represent a departure from the basic state of the fluid. For fixed value of the fluid viscosity, instability is shown to be a function of the amplitude of oscillation of the basic flow, excitation frequency, and the channel wall geometry. Special attention is given to pressure gradient and surface wave driven flows. Conditions for instability and transition are outlined.

3:10–3:25 Break

Contributed Papers

3:25

1pPA6. Examples of viscous phenomena relevant to second-order responses to ultrasound. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu) and Likun Zhang (Phys. Dept., Univ. of Texas, Austin, TX)

It has long been realized that viscosity strongly affects the response of drops and bubbles to the radiation pressure of modulated ultrasound. See for example an analysis of the viscous damping of bubble and drop shape oscillations and related experiments [T. J. Asaki and P. L. Marston, J. Fluid Mech. **300**, 149–167 (1995)]. There has been some recent interest in the importance of the viscosity of the fluid surrounding objects of interest when considering other second-order responses. These include the following quantities or responses associated with scattering by spheres and other objects: radiation force of progressive plane waves and Bessel beams [P. L. Marston, Proc. Meet. Acoust. **19**, 045005 (2013)]; power extinction of acoustic beams [L. Zhang and P. L. Marston, Bio. Opt. Express **4**, 1610–1617 (2013); (E) 4, 2988 (2013)]; and the radiation torque of vortex beams and orthogonal standing waves [L. Zhang and P. L. Marston, J. Acoust. Soc. Am. **131**, 2917–2921 (2014)]. The underlying assumptions and limitations of some current and prior approaches will be noted along with some applications to acoustophoresis. [Work supported by ONR (Marston) and by the 2013-14 F. V. Hunt Postdoctoral Fellowship (Zhang).]

3:40

1pPA7. Understanding the fluid dynamics associated with macro scale ultrasonic separators. Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA), Kedar Chitale, Walter Presz (FloDesign Sonics, 380 Main St., Wilbraham, MA 01095, k.chitale@fdsonics.com), and Olivier Desjardins (Mech. and Aerosp. Eng., Cornell Univ., Ithaca, NY)

Acoustic standing wave fields are widely used in MEMS applications to separate micron sized particles from fluids. However, the use and understanding of macro scale ultrasonic separators are still limited and challenging. These systems rely on acoustic radiation forces for trapping and clumping of dispersed phase particles. The clumps of particles then continuously separate out due to enhanced gravity or buoyancy in a flowing system. Typical flow Reynolds numbers are less than 50, particle concentrations up to 20%, ultrasonic standing wave fields at frequencies of 2 MHz, and acoustic pressure amplitudes of about 1 MPa. At such small Reynolds numbers, the flow is dominated by shear forces and the drag on clumps of particles is significantly lower than Stokes drag on a single particle. The fluid dynamics associated with these systems is extremely complex due to the coupling between the fluid flow field, suspended particles, and acoustic radiation forces. This work discusses the key physics involved and explains our current understanding of operation of macro scale acoustic separators. The status of CFD efforts to predict the flow fields and particle clumping in such systems is presented and compared to experimental results.

3:55

1pPA8. Experimental and numerical acoustofluidics in bulk acoustic wave devices at ETH Zurich. Philipp Hahn, Ivo Leibacher, Andreas Lamprecht, Peter Reichert, and Jurg Dual (Inst. of Mech. Systems (IMES), ETH Zurich, Tannenstrasse 3, Zurich CH-8092, Switzerland, hahnp@ethz.ch)

Ultrasonic fluid cavity resonances in acoustofluidic micro-devices can be exploited to miniaturize important operations for the handling of beads, cells, droplets, and other particles. With a growing number of experimentally tested unit operations, acoustofluidics holds increasing promise for emerging applications in bio- and microtechnology on lab-on-a-chip systems. We provide an overview of our research activities during the last years with a focus on the latest experimental setups and advances in the numerical simulation. Specifically, we present micro-devices with impedance matched cavity walls that allow a more flexible device design. Further, we show devices for the handling of fluid droplets and report on a method for the direct measurement of the acoustic radiation force on micro-particles. Due to the rapidly growing computational capabilities, numerical simulation has become a valuable tool in acoustofluidics research. We present a numerical model that accurately mimics the boundary layer damping inside the fluid cavity, allowing to make predictions of the attainable acoustic amplitudes and radiation forces. Furthermore, we demonstrate how numerical optimization of the device geometry is used to design new devices in an automatic fashion. Finally, we show how the radiation forces and torques can be deduced from the simulated acoustic fields to compute trajectories of complex shaped particles.

4:10

1pPA9. Inhibition of Rayleigh-Bénard convection via oscillatory acceleration. Anand Swaminathan, Steven L. Garrett (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, aswaminathan@wesleyan.edu), and Robert W. Smith (Appl. Res. Lab., The Penn State Univ., State College, PA)

The ability to dynamically stabilize Rayleigh-Bénard convection by imposition of sinusoidal acceleration is of interest to groups who design and study thermoacoustic machines, as the introduction of unwanted convection can have deleterious effects on the desired operation and efficiency of the machine. These performance issues, tentatively attributed to convective instability, have been observed both in traveling wave thermoacoustic refrigerators and in pulse tube cryocoolers. This presentation discusses an ongoing experiment designed to determine the vibratory conditions under which a small, rectangular container of statically unstable fluid may be stabilized by vertical vibration, to test the computational methods of R. M. Carbo [J. Acoust. Soc. Am. **135**(2), 654–668 (2014)]. Measurement methods developed to determine the onset and inhibition of convection will be discussed. These include the measurement of heat transport employing a feedback thermal control loop and direct optical observation using a lightsheet produced by a laser diode source illuminating seeded particles. Preliminary results in both the static and vibratory conditions will be presented. [Work supported by the ARL Walker Graduate Assistantship, the Office of Naval Research, and ARPA-E.]

4:25

1pPA10. Signal coherence of broadband sound propagation through a refractive and turbulent atmosphere. Jericho E. Cain, Sandra L. Collier (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, jericho.cain@gmail.com), Vladimir E. Ostashev, and David K. Wilson (U.S. Army Engineer and Development Ctr., Hanover, NH)

Atmospheric turbulence and refraction have significant effects on acoustic signals. The signal coherence can be directly measured and can yield information for use in source localization and classification; however, the effects of turbulence and refraction decrease the accuracy of such efforts. The signal coherence of a broadband acoustic signal above an impedance ground in a refractive, turbulent atmosphere with spatial fluctuations in the temperature and wind velocity is modeled and the results for several atmospheric cases are considered.

4:40

1pPA11. The use of sound speed in downhole flow monitoring applications. Haldun Unalmis (Weatherford, 22001 North Park Dr., Kingwood, TX 77339, haldun.unalmis@weatherford.com)

This paper describes the use of sound speed in flow monitoring applications in the high-pressure/high-temperature downhole environment. Downhole flow monitoring is an area that continuously receives attention for many reasons including zonal production allocation in multi-zone intelligent completions with inflow control valves (ICV), detection of production anomalies, as well as reduction of surface well tests and facilities. The propagation speed of a sound wave is a powerful tool to extract useful information from a flowing fluid medium in pipe whether the medium consists of a single-phase or multiphase flow. Considering the complex nature of the flow patterns and changing phase fractions from reservoir to surface, obtaining the propagation speed of sound in this harsh environment is not a trivial task, especially if the interest is real-time flow monitoring. The demanding applications span a wide spectrum from very noisy medium (usually created in gas/liquid flows by the presence of ICV) to very quiet medium, which usually originates from slow-moving liquid/liquid flows. Real-life examples are used for demonstrations. Although most examples are based on strain-based local sensing of the flow, the use of sound speed is independent of the methodology and can be implemented by other methods such as acoustic-based distributed sensing.

4:55

1pPA12. Numerical investigation of aeroacoustic characteristics of a circular cylinder in cross-flow and in yaw. Xiaowan Liu, David J. Thompson (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton SO17 1BJ, United Kingdom, xl25g11@soton.ac.uk), Zhiwei Hu (Univ. of Southampton, Southampton, United Kingdom), and Vincent Jurdic (Arup Acoust., Winchester, United Kingdom)

It is well known that aerodynamic noise becomes dominant when the speed of high-speed trains exceeds about 300 km/h. The pantograph is one of the main aerodynamic noise sources, particularly in the presence of noise barriers which shield the sources on the lower part of the train more effectively. The pantograph consists of a number of slender bodies with different cross-sections. In the current research, the aerodynamic characteristics of a circular cylinder have been investigated through computational fluid dynamics simulations using a Delayed Detached-Eddy Simulation model. Then the aeroacoustic behaviour has been predicted by using Ffowcs Williams-Hawkings equation. Simulations have been carried out for various speeds, resulting in a wide range of Reynolds number, which includes subcritical, critical, and supercritical flow states. The results have been compared with experiments and give good agreement. As the pantograph arms are inclined to the flow, the effect of yaw angle is analyzed in this paper and the effect on vortex-shedding frequency and noise level is determined. Moreover, it is demonstrated that the critical Reynolds number, which determines the beginning of the critical flow state, is affected by the yaw angle. In addition, considering the high turbulence intensity on the train roof, a turbulent inflow with different levels of intensity has been considered and the results will be presented.

Session 1pPP

Psychological and Physiological Acoustics: Psychoacoustics (Poster Session)

Nathaniel Spencer, Chair
Communication Sciences and Disorders, University of Pittsburgh, 5056 Forbes Tower, Pittsburgh, PA 15213

Posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 1:15 p.m. to 2:45 p.m., and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 4:30 p.m. There will be a 15-minute break from 2:45 p.m. to 3:00 p.m.

Contributed Papers

1pPP1. Pitch perception: Spectral and temporal integration at very high frequencies. Bonnie K. Lau and Andrew J. Oxenham (Univ. of Minnesota, 27 East River Rd, N625 Elliot Hall, Minneapolis, MN 55455, bonniekwlau@gmail.com)

Recent work has shown that complex pitch can be extracted from spectrally resolved harmonics, even when all the harmonics are higher than 6 kHz, and so unlikely to be represented by a temporal phase-locked neural code. This study measured spectral and temporal integration of frequency and fundamental-frequency (F0) discrimination at such high frequencies and compared the patterns of results to those obtained at lower frequencies. Difference between the present and earlier studies included the use of level roving on individual components to reduce loudness cues, and the careful control of audibility at very high frequencies. The low and high spectral regions consisted of harmonics 6–10 of two nominal F0s (280 and 1400 Hz). F0 difference limens (DLs) of the complexes and DLs of each harmonic individually were measured at two durations (30 and 210 ms) to measure spectral and temporal integration effects. All tones were presented in background noise to mask distortion products. Preliminary data show similar temporal integration at low and high frequencies, but very different spectral integration, with high-frequency F0DLs generally lower than predicted by a simple detection-theory model. The results suggest different integration mechanisms in the two regions. [Work supported by NIH grant R01DC05216.]

1pPP2. Effects of resolved and unresolved harmonics in the perception of multiple pitches. Jackson E. Graves and Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, grave276@umn.edu)

The mechanisms underlying pitch perception have been studied for several decades. Most studies have concentrated on the pitch of single sounds, some have investigated the pitch of two simultaneously presented sounds, but few if any have investigated pitch perception with three or more simultaneous pitches. In contrast, in music it is rule, rather than the exception, that we are presented with multiple pitches. Here, pitch perception was studied with three simultaneously presented harmonic complexes, filtered into a single bandpass region to control the extent to which harmonics from each of the three tones were spectrally resolved. The tones comprised the three possible inversions of the major or minor triad, and the listener's task was to discriminate between major and minor. Preliminary data suggest that subjects who could reliably discriminate major and minor triads in a pure tone training task were also capable of this discrimination when tone complexes were filtered such that the combination of harmonic components should have been unresolved on the cochlea. This finding suggests resolved harmonics may not be necessary to extract the pitch from multiple complex tones. Predictions from various spectral and temporal models of pitch were compared with the results. [Work supported by NIH grant R01DC05216.]

1pPP3. On the search of the optimal range and number of lags for an autocorrelation model of pitch. Sebastian Ruiz-Blais and Arturo Camacho (Universidad de Costa Rica, Guadalupe, Goicoechea, San José 1385-2100, Costa Rica, ruizble@yahoo.com)

Autocorrelation-based pitch models account for a number of pitch phenomena. Previous work has demonstrated their successfulness in response to different stimuli, when combined with a model of the auditory nerve activity and summed across channels [Balaguer-Ballester, Denham & Meddis, J. Acoust. Soc. Am. **124**, 2186–2195 (2008)]. However, autocorrelation models are generally thought of as biologically unrealistic because they require many calculations and might not tally with the neurobiological principle of economy. The present work consists of a series of simulations that aim to determine whether all operations required to compute autocorrelation are strictly necessary to account for the psychophysical data or not. To investigate this, different ranges and densities of lags are used, and the produced implementations are tested with stimuli comprising pure tones, resolved and unresolved complex tones, inharmonic complex tones, and click trains. As a result, the optimal range and number of lags is found, as the most economical setup that agrees with psychophysical evidence.

1pPP4. Relation between measures of psychoacoustic abilities and masking release for unprocessed, envelope, and temporal fine-structure speech in listeners with normal and impaired hearing. Agnes C. Leger (School of Psych., Univ. of Manchester, Ellen Wilkinson Bldg., Oxford Rd., Manchester M13 9PL, United Kingdom, agnes.leger@manchester.ac.uk), Joseph G. Desloge, Charlotte M. Reed, and Louis D. Braida (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Boston, MA)

Léger *et al.* [J. Acoust. Soc. Am. **135**, 2160, 2014] studied masking release in normal-hearing and hearing-impaired listeners using unprocessed speech and speech processed to convey envelope (ENV) or temporal fine-structure (TFS) cues. Consonant identification scores in speech-shaped background noise (continuous and square-wave interrupted at a rate of 10 Hz) indicated a substantial masking release (MR, better performance in the interrupted noise) for all speech types for normal-hearing listeners but only for the TFS speech in most hearing-impaired listeners. The current study was concerned with relating consonant identification scores and masking release to SRTs for sentences in both types of noise as well as to measurements of basic psychoacoustic abilities. These measures included absolute pure-tone detection thresholds, estimates of auditory bandwidth using a notched-noise procedure, and estimates of cochlear compression obtained using a forward-masking paradigm. For both unprocessed and ENV speech, intelligibility scores and MR were correlated with absolute thresholds and auditory bandwidths in the mid-frequency region (but not at a lower frequency). These correlations were less strong for TFS speech intelligibility and did not predict MR. Cochlear compression was not related to speech intelligibility when controlling for the effect of absolute thresholds. [Work supported by NIH R01 DC000117.]

1pPP5. Sound source similarity influences change perception during complex scene perception. Kelly Dickerson (Human Res. and Eng. Directorate, Army Res. Lab., 131 Waldon Rd., Abingdon, MD 21009, dickersonkelly23@gmail.com), Jeremy Gaston, Ashley Foots, and Timothy Mermagen (Human Res. and Eng. Directorate, Army Res. Lab., Aberdeen Proving Ground, MD)

Everyday listening involves identifying an ever-changing milieu of sound sources in the environment. Recent studies have demonstrated that change perception during complex listening tasks is highly error prone; errors can exceed 30% for sounds that are clearly detectable and identifiable in isolation. This change deafness has been generally attributed to failures of attention or memory. The current study investigates the possibility that lower-level informational effects such as acoustic or semantic similarity are predictive of change perception errors. Listeners were briefly presented with pairs of auditory scenes. In experiment 1, the listeners' were asked to identify *if* a change in sound source occurred. In experiment 2, listeners were asked to indicate *where* a change occurred across an array of loudspeakers. Results indicate that performance on the change identification and localization task was strongly influenced by the degree of similarity between the changed source and the "background" sources. Further, the semantic heterogeneity of the background seemed to be important predictor of performance on these change perception task.

1pPP6. Forward masking in children and adults with normal hearing and hearing loss. Marc A. Brennan, Walt Jesteadt, and Ryan McCreery (Res., Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68131, marc.brennan@boystown.org)

The objective was to determine the extent to which childhood hearing loss can affect the development of temporal resolution. Participants were divided into those with and without hearing loss. Participants included 16 children per group (normal hearing, hearing loss) and 14 adults per group. Forward-masked thresholds for a 2 kHz masker and target (10 ms signal delay) were measured at similar points on the dynamic range for the participants with normal hearing and hearing loss. As expected, the amount of masking decreased with age. Due to their hearing loss, the participants with hearing loss were tested at lower masker sensation levels than the participants with normal hearing. For that reason, it was expected that less masking would occur for the participants with hearing loss, and this was observed in the data. It was hypothesized that age and hearing loss would interact, such that the younger children with hearing loss would show greater masking than younger children with normal hearing. Instead, both children with normal hearing and with hearing loss showed higher masking than their adult counterparts, suggesting that the amount of masking did not interact with age and hearing status. These findings suggest that children show less efficient temporal processing.

1pPP7. Dynamic binaural sound source localization with interaural time difference cues: Artificial listeners. Liang Sun (Dept. of Elec. and Comput. Eng., The Univ. of Texas at San Antonio, San Antonio, TX), Xuan Zhong, and William Yost (Arizona State Univ., 975 S Myrtle Ave., Lattie F Coor Hall 2211, Tempe, AZ 85287, xuan.zhong@asu.edu)

When an array of two acoustic sensors is used to localize sound sources based on time differences alone, possible solutions form a cone of confusion. This study, together with a similar one for human listeners, demonstrates that azimuth/vertical planes localization of a sound source using only time difference information is feasible when self-motion measurement of the listener is available. In particular, the case of a static sound source playing that broadcasts low frequency pure tone signals was investigated. A dummy head is mounted on top of a rotating chair to mimic the head and body motion of human beings, as well as to collect audio signals. A gyroscope was mounted on top of the dummy head to collect self-motion data. A mathematical model was constructed to describe the interaural time difference (ITD) change over time, and an Extended Kalman Filter (EKF) was used to estimate the spatial angles of the sound sources with respect to the listener using the developed mathematical model and measured data. The effectiveness and robustness of the developed algorithm are shown by both the numerical and experimental results, which reveal the quick convergence of the estimated spatial angles toward their real values given noisy measured data. The possibilities of using other spatial hearing cues were also discussed.

1pPP8. On the extent of head motion in listening tests and individual measurements using different head-tracking sensors. György Wersényi (Dept. of Telecommunications, Széchenyi István Univ., Egyetem tér 1, Györ 9026, Hungary, wersenyi@sze.hu) and Jeff Wilson (Interactive Media Technol. Ctr., Georgia Inst. of Technol., Atlanta, GA)

Measurements and applications in spatial hearing research, virtual auditory displays, etc., often rely on head-related transfer functions (HRTFs) of human subjects. Individually measured HRTFs have the advantage of being more accurate in virtual localization tasks. On the other hand, the measurement and recording procedure raise several new problems such as signal-to-noise ratio issues and subject comfort. For measurements with human subjects, lots of methods are used from free heads to different head fixations methods. This study analyses the extent of head movement during measurements using different sensors and environmental conditions based on the circular angle variance, errors in yaw-pitch-roll directions and magnitude of standard deviation. Conclusive results indicate magnitudes of standard deviation of 2–8 cm and errors about 2 degrees depending on the situation as well as a preference for sitting instead of standing posture.

1pPP9. Effects of harmonic roving on pitch discrimination. Sébastien Santurette, Mathilde Le Gal de Kérangal, and Suyash N. Joshi (Hearing Systems, Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, DTU Bygning 352, Kgs. Lyngby 2800, Denmark, ses@elektro.dtu.dk)

Performance in pitch discrimination tasks is limited by variability intrinsic to listeners which may arise from peripheral auditory coding limitations or more central noise sources. The present study aimed at quantifying such "internal noise" by estimating the amount of harmonic roving required to impair pitch discrimination performance. Fundamental-frequency difference limens (F0DLs) were obtained in normal-hearing listeners with and without musical training for complex tones filtered between 1.5 and 3.5 kHz with F0s of 300 Hz (resolved harmonics) and 75 Hz (unresolved harmonics). The harmonicity of the tone complexes was varied by systematically roving the frequency of individual harmonics, which was taken from a Gaussian distribution centered on the nominal frequency in every stimulus presentation. The amount of roving was determined by the standard deviation of this distribution, which varied between 0% and 16% of the tested F0. F0DLs for resolved harmonics remained unaffected for up to 6% roving, and increased thereafter. For unresolved harmonics, performance remained stable up to larger roving values. The results demonstrate a systematic relationship between F0DLs and stimulus variability that could be used to quantify the internal noise and provide strong constraints for physiologically inspired models of pitch perception.

1pPP10. Pitch perception and frequency-following responses elicited by lexical-tone chimeras in American and Chinese adults. Fuh-Cherng Jeng (Commun. Sci. and Disord., Ohio Univ., 1 Ohio University Dr., Grover Ctr. W224, Athens, OH 45701, jeng@ohio.edu), Meng-Shih Chou, Chia-Der Lin (China Medical Univ. Hospital, Taichung City, Taiwan), John Sabol, Grant Hollister (Commun. Sci. and Disord., Ohio Univ., Athens, OH), Ching-Hua Chen (China Medical Univ. Hospital, Taichung City, Taiwan), Jessica Kenny (Commun. Sci. and Disord., Ohio Univ., Athens, OH), and Yung-An Tsou (China Medical Univ. Hospital, Taichung City, Taiwan)

Objective: Previous research has shown the usefulness of utilizing auditory chimeras (i.e., interchanging the envelope and fine structure of two different sounds) in assessing a listener's perception of the envelope and fine structure for an acoustic stimulus. However, research comparing and contrasting behavioral and electrophysiological responses to this stimulus type is scarce. *Design*: Two sets of chimeric stimuli were constructed by interchanging the envelopes and fine-structures of the rising /yi^2/ and falling /yi^4/ Mandarin pitch contours that were filtered through 1, 2, 4, 8, 16, 32, and 64 frequency banks. Behavioral pitch-perception tasks (through a single-interval, two-alternative, forced-choice paradigm) and electrophysiological responses (through the frequency-following response to voice pitch using validated scalp-recorded potential methods) were obtained from two groups of participants (native speakers of a tonal or nontonal language). *Study Sample*: Thirteen American and 13 Chinese adults were recruited. *Results*: A two-way analysis of variance showed significance ($p<0.05$) within and across the filter bank and language background factors for the behavioral measurements to the lexical-tone chimeras, while the frequency-following response demonstrated a significance across filter banks, but not for language background. *Conclusions*: Frequency-following responses to voice pitch provide supplementary information on how chimeric stimuli are processed at the brainstem level.

1pPP11. Effects of compression channel number and linked compression on masked lateralization performance in simulated bilateral cochlear implant listening. Nathaniel Spencer and Christopher Brown (Commun. Sci. and Disord., Univ. of Pittsburgh, 5056 Forbes Tower, Pittsburgh, PA 15213, njs64@pitt.edu)

Bilateral cochlear implants (CIs) are becoming increasingly common. It is hoped for bilateral CI users that audibility in quiet will be complemented by performance benefits in everyday binaural listening tasks, like location-identification of a speech talker presented amid other talkers. While normal-hearing (NH) listeners have typically performed well in such kinds of tasks, bilateral CI users have performed poorly. It is possible that, for bilateral CI users, better performances can be attained through a re-thinking of device signal-processing strategies. In particular, we are interested in evaluating alternatives to single-channel independent compression, as is typically applied at the automatic gain control (AGC) stage. Specifically, we test the hypothesis that for simulated bilateral CI users (listeners with both NH and hearing impairments, HI), lower rms-error in a masked lateralization task will be achieved for multi-channel AGC and linked AGC, than for single-channel independent AGC, with and without pre-emphasis filtering, and for low and high compression thresholds. Preliminary data suggest a lower rms-error for linked compression than for independent compression, in conjunction with no detrimental effect of multi-channel compression. These data support the possibility of linked AGC with multi-channel AGC as an alternative to independent AGC in bilateral CI listening.

1pPP12. The build-up and resetting of auditory stream segregation: Effects of timbre, level, and abrupt change. Saima L. Rajasingam, Robert J. Summers, and Brian Roberts (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, rajasisl@aston.ac.uk)

Two experiments explored the effects of timbre, level, and abrupt change on the dynamics of streaming. Listeners continuously monitored 20-s ABA-sequences (1-kHz base frequency; $\Delta f = 4$–8 semitones) and reported when the sequence was heard as integrated or segregated. Experiment 1 used pure tones and narrowly spaced (± 25 Hz) tone pairs (dyads); both evoke similar excitation patterns but dyads have a “rougher” timbre. Dyad-only sequences induced a strongly segregated percept, with limited scope for further build-up. Abrupt alternations every 13 triplets produced large changes in the extent of segregation. Dyad-to-pure transitions produced substantial resetting, but pure-to-dyad transitions elicited greater segregation than for the corresponding time in dyad-only sequences (overshoot). Experiment 2 examined the effect of varying triplet level. Pure-tone sequences with a maintained 12-dB difference showed similar build-up, but an asymmetry occurred for alternating sequences. Increased-level transitions caused a significant reduction in segregation (resetting) whereas decreased-level transitions had little or no effect. The results suggest that timbre can strongly affect the likelihood of stream segregation, even without peripheral-channeling cues, but moderate level differences do not. Furthermore, abrupt changes in sequence properties have variable effects on the tendency to report segregation—including resetting, overshoot, and asymmetries in the effect of transition direction.

1pPP13. Individual differences in acoustic acuity at supra-threshold level. Lengshi Dai and Barbara G. Shinn-Cunningham (Dept. of Biomedical Eng., Boston Univ., 677 Beacon St., Boston, MA 02215, ldai@bu.edu)

The ability to detect acoustic amplitude modulated envelopes differs even among listeners with normal hearing thresholds (NHTs), an effect that to some degree may correlate with noise exposure history. Moreover, at least one animal study suggests that auditory nerve fibers (ANFs) with lower spontaneous firing rates (high thresholds) are more vulnerable to damage from noise exposure compared to those with high spontaneous rate ANFs. Given that NHTs depend primarily on high-spontaneous rate fibers, even relatively severe loss of low-spontaneous ANFs could be present in listeners that have “normal hearing” according to traditional hearing screenings. Here, we evaluated the robustness of supra-threshold coding, which may relate to low-spontaneous ANFs health, in listeners with NHTs. Specifically, we measured the phase-locking-values (PLVs) of subcortical envelope following responses to SAM tones (carrier: 1500 Hz; modulation rate: 100 Hz) at two sensation levels (20/50 dB SL) and with two modulation depths (0/−6 dB). Results reveal huge individual differences in PLV strengths that are negatively correlated with behavioral amplitude modulation detection thresholds for noises at a moderate supra-threshold level (carrier: band-pass-filtered noise centered at 1500 Hz with 300 Hz bandwidth; modulation rate:10 Hz). Our data support the hypothesis that the number of operational low-spontaneous rate ANFs may differ human listeners with NHTs, and that deficit in low-spontaneous rate ANFs can lead to deficits in perceptual abilities.

1pPP14. Effects of active and passive hearing protective devices on sound source localization, tone detection, and speech recognition. Andrew D. Brown, Brianne T. Beemer, Nathaniel T. Greene, and Daniel J. Tollin (Physiol. & Biophys., Univ. of Colorado School of Medicine, 12800 East 19th Ave., RC1-N, Rm. 7401G, M.S. 8307, Aurora, CO, Andrew.D.Brown@ucdenver.edu)

Hearing protective devices (HPDs) such as earplugs and earmuffs offer to mitigate noise exposure and thus noise-induced hearing loss among persons frequently exposed to intense sound, e.g., military personnel or industrial workers. However, distortions of spatial acoustic information and attenuation of low-intensity sounds caused by many existing HPDs can make their use untenable in high-risk environments where auditory situational awareness is imperative. Here, we assessed (1) sound source localization accuracy using a head-turning paradigm, (2) tone detection thresholds using a two-alternative forced-choice task, and (3) speech-in-noise recognition using a modified version of the QuickSIN test in 10 young normal-hearing males wearing four different HPDs (two active, two passive), including two new and previously untested devices. Relative to unoccluded (control) performance, all tested HPDs significantly degraded performance across tasks, although one active HPD slightly improved high-frequency tone detection thresholds, and did not degrade speech recognition. Behavioral data were examined with respect to binaural acoustic information (directional transfer functions) measured in a binaural manikin with and without tested HPDs. Data reinforce previous reports that HPDs significantly compromise auditory perceptual facilities, particularly sound localization due to distortions of high-frequency pinna cues.

1pPP15. Sequential streaming under reverberation: Effects of the reverberant tail properties. Eugene J. Brandewie and Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, 75 E. River Rd., Minneapolis, MN 55455, ebrandew@umn.edu)

A physical difference between two alternating stimuli can elicit perceptual segregation when the difference is sufficiently salient. One such cue involves differences in reverberation, potentially caused by differences in source distance. Here, we studied what aspects of difference in reverberation are most important in eliciting segregation using a rhythmic masking task. Two interleaved sequences of Gaussian noise bursts (target and interferer) were presented on each trial and listeners attempted to identify which of two rhythms was presented in the target sequence. The influence of the reverberation tail (or damped decay) was studied by parametrically changing its duration in the target sequence, while eliminating all binaural cues. The influence of spectral content of the tail was examined by simulating the spectral coloration produced by real rooms. Results suggest that damped tails can elicit perceptual segregation with tail durations less than 100 ms. In addition, the spectral content of the tail can further influence segregation performance. Overall, differences in reverberation can serve as a prominent cue to aid perceptual segregation, particularly if a room environment introduces differences in spectrum based on distance, which is often the case in real rooms. [Work supported by NIH grant R01DC07657.]

1pPP16. Streaming and sound localization with a preceding distractor. Gabriela Andrejkova (Inst. of Comput. Sci., P. J. Safarik Univ. in Kosice, Jesenna 5, Kosice 04001, Slovakia, gabriela.andrejkova@upjs.sk), Virginia Best, Barbara G. Shinn-Cunningham (Boston Univ., Boston, MA), and Norbert Kopčo (Inst. of Comput. Sci., P. J. Safarik Univ. in Kosice, Kosice, Slovakia)

A previous study of sound localization with a preceding distractor showed that (1) the distractor affects response bias and response variance for distractor-target inter-stimulus-intervals of up to 400 ms, and that (2) localization responses are biased away from the distractor even on interleaved control trials in which the target is presented alone [Kopco *et al*., JASA **121**, 420–432, 2007]. Neural mechanisms operating on time scales of milliseconds to tens of seconds are likely to cause to these effects. The current study examined how perceptual organization affects target localization performance. Sound localization was examined for 2-ms click target stimuli. On 80% of trials, the target was preceded by a distractor, designed either to be grouped with the target (distractor was an identical 2-ms click) or to be perceived in a separate stream (an isochronous train of eight clicks whose inter-click-interval was different from the distractor-target inter-stimulus-interval). As hypothesized, the single-click distractor affected target localization more than the eight-click distractor. On the other hand, the biases in the control trials were greater for the eight-click distractor. These results indicate that performance is influenced by both top-down mechanisms like streaming and bottom-up mechanisms like stimulus distribution-based adaptation. [Work supported by APVV-0452-12 and R01DC009477.]

1pPP17. Multiple sound source localization when sounds are stationary or rotating. Xuan Zhong and William Yost (Arizona State Univ., 975 S Myrtle Ave., Lattie F Coor Hall 2211, Tempe, AZ 85287, xuan.zhong@asu.edu)

In the field of spatial hearing, it is known that the sound source localization error increases when distractors are present. This study first determined the maximum number of sound sources that can be perceived. In one experiment, listeners determined how many sources (loudspeakers) presented sounds and these loudspeaker locations. In another experiment, listeners were asked to locate a new sound source which was added to an existing set of fixed sources. Listeners determined up to about four sources producing simultaneous unrelated speech sounds in both experiments. A second part of the research examined the effect of rotating the sound sources on the listener's perception. If multiple unrelated sounds such as speech rotated, listeners could much more easily determine sound rotation than if the sounds were related to each other. If the sounds were related (e.g., all sounds were harmonics of a fundamental) but were still presented from a number of different loudspeakers, listeners could hardly process the individual sounds even when sounds rotated among the loudspeakers. Thus, the ability to locate multiple sound sources depends on the total number and motion of sources and the type of sound. [Research supported by an AFOSR grant.]

1pPP18. The addition of a second lag to the lead-lag precedence effect paradigm for temporally overlapping noise stimuli. M. Torben Pastore and Jonas Braasch (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., 4 Irving Pl., Troy, NY 12180, m.torben.pastore@gmail.com)

In reverberant conditions, humans routinely perceive sound sources in the direction of the first wavefront despite competing directional information presented by a host of reflections arriving soon after—the so-called "Precedence Effect." This is often tested over headphones using a "direct sound" (lead) with a single delayed copy (lag) serving as a modeled reflection. Previously, we employed this common experimental paradigm to investigate the lateral extent of the precedence effect using temporally overlapping noise stimuli. The current study extends this inquiry towards the multiple reflections encountered in room-acoustic scenarios by presenting a second lag. Lead and lag stimuli are 200-ms Gaussian noise (500-Hz center frequency, 800-Hz bandwidth) presented dichotically with a programmable amount of delay for both lags. Relative to the intensity of the lead, the two lags are presented at 0, -3, and -6 dB. The lead is presented at the midline, with an ITD of 0 μs. The two lags are delayed by between 1 and 7 ms, with opposing ITDs of ± 300 μs. Listeners indicate the lateralization of their auditory event with an acoustic pointer.

1pPP19. Relation between temporal envelope coding, pitch discrimination, and compression estimates in listeners with sensorineural hearing loss. Federica Bianchi, Sébastien Santurette, Michal Fereczkowski, and Torsten Dau (Tech. Univ. of Denmark, Ørsteds Plads Bldg. 352, Lyngby, Denmark 2800, Denmark, fbia@elektro.dtu.dk)

Recent physiological studies in animals showed that noise-induced sensorineural hearing loss (SNHL) increased the amplitude of envelope coding in single auditory-nerve fibers. The present study investigated whether SNHL in human listeners was associated with enhanced temporal envelope coding, whether this enhancement affected pitch discrimination performance, and whether loss of compression following SNHL was a potential factor in envelope coding enhancement. Envelope processing was assessed in normal-hearing (NH) and hearing-impaired (HI) listeners in a behavioral amplitude-modulation detection task. Fundamental frequency difference limens (F_0DLs) were obtained in the same listeners for complex tones with varying harmonic resolvability. Basilar-membrane input/output functions were measured to assess individual compression ratios. For NH listeners, F_0DLs decreased with increasing harmonic resolvability. For the unresolved conditions, all five HI listeners performed as good as or better than NH listeners with matching musical experience. Two HI listeners showed lower amplitude-modulation detection thresholds than NH listeners for low modulation rates, and one of these listeners also showed a loss of cochlear compression. Overall, these findings suggest that some HI listeners may benefit from an enhancement of temporal envelope coding in pitch discrimination of unresolved complex tones, and that this enhancement may be also ascribed to a reduction of cochlear compression following SNHL.

1pPP20. Time dependence of the time interval in distance/ intensity threshold in human listeners. Larisa Dunai, Ismael Lengua, and Miguel Iglesias (Universitat Politècnica de València, St. Camino de Vera s/n 5L, Valencia 46022, Spain, ladu@upv.es)

The paper presents the distance threshold for complex virtual sounds localization in dependence of the time interval. The method of measurement was a forced-choice, three-down one-up staircase. The sound is a complex record of a click sound of 47 ms at 44.1 kHz measured by using a maximum length binary sequence in an anechoic chamber. The azimuth of the experimental is 0°, at the center of the human head where the ITD = 0, varying just the intensity. The investigation reported that the lowest distance thresholds of 0.06 m occurred for high inter click intervals at 200 ms–300ms. At lower ICIs, the distance threshold increases considerably. The experimental was carried out with the objective to find the distance threshold and time interval for implementing the data for a navigation device for blind people.

1pPP21. Time-efficient multidimensional threshold tracking method. Michal Fereczkowski, Borys Kowalewski, Torsten Dau, and Ewen N. MacDonald (Elektro, DTU, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, mfer@elektro.dtu.dk)

Traditionally, adaptive methods have been used to reduce the time it takes to estimate psychoacoustic thresholds. However, even with adaptive methods, there are many cases where the testing time is too long to be clinically feasible, particularly when estimating thresholds as a function of another parameter, such as in temporal masking curves or when characterizing auditory filters. Here we present a new method, the "grid" method, which adaptively varies multiple parameters during each experimental run. By changing the way the parameter-response space is sampled, the method increases the proportion of experimental time spent in the vicinity of the sought-after threshold curve. The resulting increase in time-efficiency is substantial and can make some measurements clinically feasible. Thresholds from temporal masking curves obtained with the grid method are compared with those from one of the most time-efficient standard methods (single-interval up-down adaptive method of Lecluyse, 2013). Overall, individuals' results from both methods are very highly correlated, but the grid method was an order of magnitude faster in estimating thresholds. The application of the grid method to other measurements, such as characterizing auditory filters, will also be discussed.

1pPP22. Measuring rapid adaptation to complex acoustic environments in normal and hearing-impaired listeners. Sofie Aspeslagh (MRC/CSO Inst. of Hearing Res.—Scottish Section, Glasgow Royal Infirmary, 16 Alexandra Parade, Glasgow G31 2ER, United Kingdom, sofie@ihr.gla.ac.uk), Fraser Clark (School of Eng. and Computing, Univ. of the West of Scotland, Paisley, United Kingdom), Michael A. Akeroyd, and W. O. Brimijoin (MRC/CSO Inst. of Hearing Res. – Scottish Section, Glasgow, United Kingdom)

Listeners can move from room to room while conversing, encountering changes in the acoustics of their surroundings. In order to maintain speech intelligibility when faced with changing noises and reverberation listeners may rapidly adapt by building and updating a model of their surrounding auditory space. We have previously demonstrated rapid adaptation to a change in the acoustic environment, consisting of a brief decrease and then a recovery in intelligibility. In the current study we investigated whether the amount and time-course of this adaptation changes as a function of age and/or hearing impairment, and whether the amount of reverberation or the complexity of the background noise would affect adaptation time. We asked 29 normal-hearing and 37 hearing-impaired listeners to identify an ongoing stream of random target words, presented at a rate of one every 1.5 seconds, while the acoustic environment was switched every nine seconds. On average, intelligibility dropped by 16% upon entering a new environment before recovering to ceiling performance within 2.3 seconds. These results were not affected by hearing impairment or age. Preliminary analyses suggest that listeners may require additional time to adapt noises of greater complexity. [Work supported by MRC (U135097131) and the Chief Scientist Office (Scotland).]

1pPP23. An adaptive procedure for estimating the auditory-filter phase response. Niall A. Klyn 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)

Current techniques for psychophysical estimates of the phase curvature of the basilar membrane rely on laborious data collection. For each point on the basilar membrane a separate threshold estimate — more commonly several estimates — must be found for each phase relationship. This practically limits the precision of the estimates and the number of subjects for whom estimates may be obtained. Here, we propose a rudimentary adaptive technique for rapidly estimating the phase curvature of the basilar membrane psychophysically. Consistent with the extant data the procedure assumes that the underlying threshold versus masker curvature function has a U shape. Rather than find thresholds at each value of masker curvature, the procedure seeks out only the value which produces the least masking. It does so by dropping the signal level until only three adjacent masker curvature values fail to mask the signal. New upper and lower boundaries for the masker curvature are then set, the step size between masker curvature values is reduced, and the previous steps are repeated. We present a comparison of the estimates produced by the conventional techniques and the proposed adaptive technique, as well as their relative efficiencies.

1pPP24. Measuring the auditory phase response based on interaural time differences. Hisaaki Tabuchi, Bernhard Laback, Piotr Majdak, Thibaud Necciari, and Katherina Zenke (Austrian Acad. of Sci., Acoust. Res. Inst., Wohllebengasse 12-14, Vienna 1040, Austria, tabuchi@kfs.oeaw.ac.at)

Harmonic complexes are often used as maskers for measuring the cochlear phase response. The phase curvature can affect the masking in order of up to 20 dB, an effect known as the masker-phase effect. There is evidence that signals yielding peaky internal masker representations after passing the cochlear filter produce minimum masking, with the fast-acting cochlear compression as the main contributor to that effect. Thus, in hearing-impaired listeners showing reduced or absent compression, the estimation of phase response using the masking method may be difficult. Here, an alternative method is proposed, which is based on the effect of signal peakedness on the sensitivity to interaural time differences (ITD) in the signal envelope. With the two methods, ITD and masking thresholds were measured, respectively, in seven normal-hearing listeners. The stimuli were 300-ms Schroeder-phase harmonic complexes, ranging from 3400 to 4600 Hz with a 100-Hz fundamental frequency, with the signal phase curvature varied between −1 and 1. The lowest ITD thresholds were observed for phase curvatures that produced minimum masking. This suggests that the ITD method is a promising candidate for estimating the cochlear phase response in hearing-impaired listeners.

1pPP25. Cochlear fine structure predicts behavioral decision weights in a multitone level discrimination task. Jungmee M. Lee (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, 1410 E. Skyline Dr., Madison, WI 53705), Glenis Long (Speech, Lang. and Hearing Sci., Graduate Ctr., City Univ. of New York, New York, NY), Inseok Heo, Christophe Stoelinga, and Robert Lutfi (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, Madison, WI, ralutfi@wisc.edu)

Listeners show highly replicable, idiosyncratic patterns of decision weights across frequency affecting their performance in multi-tone level discrimination tasks. The different patterns are attributed to peculiarities in how listeners attend to sounds. However, evidence is presented in the current study that they reflect individual differences in cochlear micromechanics, which can be evaluated using otoacoustic emissions (OAEs). Spontaneous OAEs (SOAEs) and the fine structure of stimulus-frequency OAEs (SFOAEs) were measured in a group of normal-hearing listeners. The same group of listeners performed a two-tone, sample-level discrimination task wherein the frequency of one tone was selected to correspond to a SOAE and the other was selected well away from a SOAE. Tone levels were either 50 or 30 dB SPL. The relative decision weight of the two tones for each listener and condition was estimated from a standard COSS analysis of the trial-by-trial data [Berg (1989), J. Acoust. Soc. Am. **86**, 1743–1746]. A strong linear relation was observed between the average relative decision weight and the average relative level of both SOAE and SFOAE.

1pPP26. A meta-analysis of the effects of hearing impairment and hearing aids on directional hearing. Michael Akeroyd and William M. Whitmer (MRC/CSO Inst. of Hearing Res. - Scottish Section, New Lister Bldg., Glasgow Royal Infirmary, Glasgow, Strathclyde G31 2ER, United Kingdom, maa@ihr.gla.ac.uk)

We report here meta-analyses of all the experiments that have been done since the 1980s on how accurate hearing-impaired listeners are at determining the spatial direction of sound sources. The results demonstrate that their performance is somewhat worse than normal hearing listeners for all directions, and especially so for sounds presented from the side (including distinguishing front vs. back) or for changes in elevation. There is considerable variation across listeners and experiments. In general, hearing aids do not improve performance, and there is overall little effect of differences in hearing aid features or designs (e.g., for left/right accuracy, an across-experiment mean difference of just 1 degree of RMS error). In many situations, a unilateral fitting results in a localization deficit. Though statistically significant effects of aided localization have been observed experimentally, few of them are generalizable as they often occurred for just some source directions, stimuli, or groups of listeners. Overall, there is no experimental evidence for a substantial, universal benefit from hearing aids to directional accuracy.

1pPP27. Effect of spectrally-remote maskers on sentence recognition by adults and children. Carla L. Youngdahl, Sarah E. Yoho, Rachael Frush Holt, Frederic Apoux, and Eric W. Healy (Speech & Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, carla.youngdahl@gmail.com)

Adults display improved detection of a signal in noise when the spectral frequency of that signal is known, relative to when it is unknown. In contrast, infants do not display this improvement, suggesting that they monitor all frequencies equally, even when it is not advantageous to do so. To assess the impact of this "spectral attention" development during speech recognition, sentences in noise were lowpass filtered at 1500 Hz and presented along with spectrally remote low-noise noise maskers that produced no spectral overlap of peripheral excitation. As anticipated, sentence recognition by adults was not affected by the presence of remote maskers, irrespective of their bandwidth and spectral location. This result was also observed

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in a group of 7-year-old children. However, the youngest children tested (5-year-olds) displayed poorer sentence recognition in the presence of the remote maskers, suggesting that they were unable to focus attention on the spectral region of speech. The current results suggest that an important step occurs in the development of auditory spectral attention around age 6. These results may also help explain why children typically require more favorable signal-to-noise ratios than adults to achieve similar levels of speech performance in noise. [Work supported by NIH.]

1pPP28. Adult listener's detection thresholds for vowels and two-syllable words in different two-talker maskers. Monika-Maria Oster and Lynne A. Werner (Speech and Hearing Sci., Univ. of Washington, 1417 North East 42nd St., Seattle, WA 98105, mmoster@uw.edu)

There is an increased interest in using natural speech targets and speech maskers to assess infants' auditory skill development. While providing ecological validity, natural speech is an inherently uncontrolled signal. Discrepancies between existing studies may be the result of differences in the maskers and/or target stimuli used. This study explores adult listener's detection thresholds for vowels and two-syllable words in three different two-talker maskers to assess the impact of (1) masker characteristics and (2) differences in target signals. Each two-talker masker was created by recording two female speakers reading different passages in monotone, in adult-directed style, or infant-directed style. Silences longer than 300 ms were deleted, and the passages' root-mean-square amplitudes balanced before combining them. The target stimuli consisted of the vowels /a/, /o/, /e/ and the words "baby" and "ice-cream." They were spoken by different female talkers in both adult-directed and infant-directed styles. An adaptive two-interval forced choice method was used to estimate threshold. Preliminary results suggest that there are small differences in detection thresholds between the different two-talker maskers. Larger differences are found between the thresholds of different target signals, indicating that discrepancies between existing studies may be due to differences in target stimuli used.

1pPP29. Effect of listener experience on pitch and rhythm perception with tonal sequences. Sandra J. Guzman, Cody Elston (Audio Arts & Acoust., Columbia College Chicago, 33 East Congress Parkway, Ste. 600, Chicago, IL 60605, sguzman@colum.edu), Valeriy Shafiro, and Stanley Sheft (Communications Disord. & Sci., Rush Univ., Chicago, IL)

Experience and training can influence discrimination of tonal sequences. Current work investigated pitch and rhythm processing of four-tone sequences by audiology and speech students, audio-arts students experienced in critical listening, and trained musicians. Sequence tones either had a fixed duration (212 ms) with frequency randomly selected from a logarithmically scaled distribution (400–1750 Hz), a fixed frequency (837 Hz) with a randomly selected log scaled duration (75–600 ms), or a random frequency and duration. In initial conditions, the task was to assemble sequence elements to recreate the target sequence for each of the three sequence types. To evaluate effect of extraneous randomization, both frequency and duration were randomized in the final two conditions with only one of the two attributes defining the target sequence. Audio-arts students performed significantly better than audiology and speech students in the reconstruction task. In conditions involving joint processing of sequence pitch contour and rhythm, the performance of audio-arts students was well approximated by the optimal combination of uncorrelated but integral stimulus dimensions. Ongoing work is evaluating the performance of trained musicians in the sequence-reconstruction task, with emphasis on manner in which information is combined across the dimensions of sequence pitch and rhythm. [Work supported by NIH.]

1pPP30. Fast frequency selectivity measures in listeners with severe hearing loss. Eric C. Hoover (Univ. of South Florida, 3802 Spectrum Blvd., Ste. 210A, Tampa, FL 33612, erichoover@usf.edu), Michael C. Blackburn (Captain James A. Lovell Federal Health Care Ctr., North Chicago, IL), and Pamela E. Souza (Northwestern Univ., Evanston, IL)

Communication difficulties in severe hearing loss are a combination of elevated pure-tone thresholds and suprathreshold deficits; for example, frequency selectivity, which may be substantially impaired in listeners with severe loss. The purpose of this study was to investigate frequency selectivity measures in listeners with severe loss (pure-tone average 60–90 dB HL). Two tests were used: fast psychophysical tuning curves (PTC), in which an intermittent tone was detected during a narrowband noise masker swept continuously in frequency; and spectral ripple (SR), in which the number of sinusoidal ripples in the spectrum of a broadband noise was tracked in an adaptive forced-choice procedure. Results showed that both tests have advantages when evaluating listeners with severe loss. Fast PTC provided a direct representation of tuning in a specific frequency region but was limited by the residual dynamic range of the listener. Dynamic range limitations were partially overcome by combining data from multiple trials. SR was able to be performed by listeners with minimal dynamic range but the interpretation of results was limited by potential differences in hearing across the bandwidth of the stimuli. Both fast PTC and SR revealed information about suprathreshold deficits that could help guide audiological intervention. [Work supported by NIH.]

1pPP31. Virtual auditory display validation using transaural techniques. Pavel Zahorik (Heuser Hearing Inst. and Div. of Communicative Disord., Dept. of Surgery, Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu) and Zijiang J. He (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Validation of headphone-based virtual auditory display technology is inherently difficult, given the potential for display hardware to interfere with normal sound field listening conditions due to occlusion of the pinna. This difficulty is eliminated if loudspeaker-based transaural techniques are instead used for the virtual auditory display. Transaural techniques also offer the advantage of being able to use the display loudspeakers, or other additional loudspeakers, as individual reference locations in order to validate the display under natural listening conditions in the same sound field. Such validation can also include explicit comparison to visual localization of the sound sources. In this way, the accuracy and precision of virtual source localization can be directly compared to real source localization, and optionally to visual localization. Results of a validation experiment of this type using non-individualized head-related transfer functions are reported and compared to analogous data from a headphone-based virtual auditory display. [Work supported by NEI.]

1pPP32. Application of a monaural glimpsing model to binaural speech mixtures. Virginia Best, Christine R. Mason, Jayaganesh Swaminathan, Elin Roverud, and Gerald Kidd (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, cmason@bu.edu)

Under certain circumstances, listeners with sensorineural hearing loss demonstrate poorer speech intelligibility in spatially separated speech maskers than those with normal hearing. One important issue in interpreting these results is whether the target speech information is available or audible in the spatialized mixture. Simple energy-based "glimpsing" models can be used to quantify the available target in speech mixtures. For example, as target level decreases in a fixed-level masker, fewer and fewer good glimpses are available. Moreover, those glimpses may not all be above threshold, particularly for individuals with hearing loss. In this study we used a glimpsing model to isolate available target glimpses in binaural speech stimuli (separately for each ear) as a function of target-to-masker ratio. Performance for stimuli processed in this way was compared to binaural performance in listeners with normal hearing and listeners with hearing loss. Results suggest that binaural performance was partly limited by the ability to use the available glimpses. Increasing the level of the glimpses gave mixed results, which will be discussed in terms of audibility. [Work supported by NIH/NIDCD.]

1pPP33. Effects of hearing impairment on sensitivity to dynamic spectral change. Michelle R. Molis, Nirmal Srinivasan, and Frederick J. Gallun (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, michelle.molis@va.gov)

The loss of peripheral auditory sensitivity, precise temporal processing, and frequency selectivity associated with hearing loss suggests that the results obtained for pure tone glide stimuli will not necessarily correspond to results obtained with more complex dynamic stimuli for listeners with hearing impairment. Normally hearing (NH) and hearing-impaired (HI) listeners identified changes in frequency as rising or falling both for tone glides and for spectrotemporal ripples. Tones glided linearly up or down in frequency with an extent of 1, 0.66, or 0.33 octaves centered around 500 or 1500 Hz. Ripple stimuli, presented in octave bands centered around 500 or 1500 Hz or in a broadband condition extending from 20–20,000 Hz, had a spectral density of 2 cycles/octave and temporal modulation gliding up or down at rates of 1, 4, or 16 Hz. Sensitivity to dynamic changes was assessed as percent correct direction identification and bias was characterized as the ratio of correctly-identified rising versus falling glides. Substantial individual variability was observed for both measures for both the NH and HI listeners, and the ability to perform the tone glide task was not a consistent predictor of the ability to perform the spectrotemporal ripple identification. [Work supported by NIH/NIDCD.]

1pPP34. Duration and transition effects in multiple-burst, multitone masking. Eric R. Thompson (Ball Aerosp. & Technologies Corp., 2610 7th St, Bldg. 441, Wright Patterson, OH 45433, eric.thompson.ctr@wpafb.af.mil), Matthew G. Wisniewski, Nandini Iyer, and Brian D. Simpson (Air Force Res. Lab, Wright-Patterson AFB, OH)

Previous studies of tone detection using a multiple-burst same (MBS) or multiple-burst different (MBD) masker have shown that tone detection thresholds were lower with an MBD masker than with an MBS masker, and decreased as the number of bursts increased. In those studies, as the number of bursts increased, the total signal duration increased as well as the number of times the (MBD) masker changed frequencies. The present study was designed to differentiate between the effects of signal duration and masker transitions. In one condition, the burst duration was inversely varied with the number of bursts so that the overall stimulus duration was always 480 ms. In a second condition, the stimuli always had eight 60-ms bursts, but the number of times the masker transitioned from one random draw of frequencies to a different draw was varied from zero (MBS) to seven (MBD). The MBD condition showed a steady improvement with number of bursts, regardless of the burst duration. There was a similar improvement with the number of masker transitions. There was only a small improvement in thresholds with increasing burst duration. Results suggest that the number of changes of masker frequencies drives performance more than the total signal duration.

1pPP35. Using multidimensional scaling techniques to quantify binaural squelch. Gregory M. Ellis (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY 40292, g.ellis@louisville.edu), Pavel Zahorik (Heuser Hearing Inst. and Div. of Communicative Disord., Dept. of Surgery, Univ. of Louisville, Louisville, KY), and William M. Hartmann (Dept. of Phys. and Astronomy, Michigan State Univ., East Lansing, MI)

Binaural squelch is a perceptual phenomenon whereby the subjective strength of reverberant sound is attenuated under binaural listening conditions relative to monaural or diotic listening conditions. Although the effect is well known, only a few studies have explicitly quantified the effect and all used uni-dimensional objective measures such as word recognition or echo detection. Here, multidimensional scaling techniques were used to more directly quantify the perceptual attributes of binaural squelch. In a series of experiments, listeners were asked to rate the perceptual similarity between pairs of sound sources that varied in distance (1–12 m), level (± 3 dB), and presentation mode (binaural versus diotic). Sounds were presented using virtual auditory space techniques to simulate sound-field listening to a frontal source in a reverberant room (broadband T60 = 2 s) and in anechoic space. The source signal was a brief sample of speech from a female talker. Multidimensional scaling (INDSCAL) results revealed that effects of binaural squelch are most evident on a perceptual dimension strongly related to the ratio of direct-to-reverberant sound energy, although individual differences were observed. Additional effects, unrelated to sound level, were also evident on a second perceptual dimension.

1pPP36. Measures of ear lateralization in a dichotic sample discrimination task. Alison Tan and Bruce Berg (Cognit. Sci., UC Irvine, 2201 Social & Behavioral Sci. Gateway Bldg., Irvine, CA 92617, aytan@uci.edu)

This study investigates the ability to lateralize using a dichotic sample discrimination task. On each trial, seven 60-ms tones are drawn from a normal distribution with means of 1000 or 1100 Hz. Even numbered tones are the most informative ($d' = 2$) and presented to one ear and the less informative, odd numbered tones ($d' = 0.5$) are presented to the other ear. Participants indicate from which distribution the tones are sampled. Task difficulty is manipulated by presenting odd-and even-numbered tones at different intensities. In easier conditions, informative and less informative tones are presented at 70 dB, 50 dB, respectively. In difficult conditions, informative and less informative tones are presented at 50 dB, 70 dB, respectively. Decision weights, efficiency measures, and performance level (d') all show an unexpectedly wide range in a listener's ability to lateralize and attend to the most informative tones. Estimates of d' range from 2.4 to 0.7. Some listeners display high efficiency estimates in all conditions while others show a marked ear preference, achieving high efficiency only when the informative tones are presented to a particular ear. Another group of listeners show a distinct inability to lateralize in any condition. This latter group is most affected by the intensity manipulation.

Session 1pSAa

Structural Acoustics and Vibration, Noise, and Signal Processing in Acoustics: Blind Source Localization and Separation for Structural Acoustic Sources

Piervincenzo Rizzo, Chair
University of Pittsburgh, 942 Benedum Hall, 3700 O'Hara Street, Pittsburgh, PA 15261

Chair's Introduction—1:00

Invited Papers

1:05

1pSAa1. Source localization and signal extraction using spherical microphone arrays. Mingsian R. Bai and Yueh Hua Yao (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan 30013, Taiwan, msbai63@gmail.com)

Source localization and extraction using spherical microphone arrays is presented in this paper. Both freefield and solid spherical arrays are examined. To minimize the basis mismatch problem in signal extraction, source localization is executed first using two beamforming methods, delay and sum (DAS) and minimum variance distortionless response (MVDR). The beamformers are formulated in both the spatial domain and the modal domain. After source localization, Tikhonov regularization (TIKR) and compressive sensing (CS) are used to extract the source amplitude signals. Simulations and experiments are conducted for spherical arrays of radius 5 cm with 32 microphones mounted on the vertices and faces of an icosahedron. The results demonstrate better localization performance of the spatial beamformer that the modal beamformer. In addition, the source speech signals are extracted by using the proposed arrays with superior quality.

1:30

1pSAa2. Integrating Markov random fields and model-based expectation maximization source separation and localization. Michael I. Mandel (Comput. Sci. and Eng., The Ohio State Univ., Columbus, OH) and Nicoleta Roman (The Comput. Sci. and Eng., The Ohio State Univ., Lima, 4240 Campus Dr., Lima, OH 45804, roman.45@osu.edu)

Separation of multiple sources from binaural reverberant audio mixtures has been shown to be a challenging task. Binaural algorithms typically perform separation by employing localization based clustering or classification in individual time-frequency units. By assuming that a single source is active in a particular time-frequency unit, these algorithms produce spectral masks that allow the separation of an arbitrary number of sources. The model-based EM Source Separation and Localization (MESSL) algorithm increases robustness to reverberation by explicitly modeling its effects on interaural phase and level differences and using a consistent estimate of interaural time difference across all frequencies. The current study has extended the MESSL algorithm with a Markov Random Field based mask smoothing procedure that enforces consistency in the assignment of neighboring time-frequency units to sources. The proposed MESSL-MRF algorithm is tested in highly reverberated conditions and shows significant improvements over the original MESSL algorithm as measured by both signal-to-distortion ratios as well as a speech intelligibility predictor.

1:55

1pSAa3. Impulsive sound localization using time-domain beamformer. Dae-Hoon Seo, Jung-Woo Choi, and Yang-Hann Kim (School of Mech., Aerosp. and System Eng., Korea Adv. Inst. of Sci. and Technology(KAIST), 373-1 Guseong-dong, Yuseong-gu, Daejeon 305-701, South Korea, ihuny@kaist.ac.kr)

This paper presents a beamforming technique for locating impulsive sound source in low signal to noise ratio (SNR). Impulsive sound generates a high peak-pressure for a short duration in time-domain. However, the measured signal of the sensor are always embedded in the interference noise sources and measurement noise, therefore, it is difficult to estimate the direction of arrival of impulse sound in low SNR environment. In contrast to a frequency-domain beamformer, it has been reported that a time-domain beamformer can be better suited for transient signals. We proposes peak and crest factor as alternative directional estimators to enhance the performance of a time-domain beamformer in view of the fact that impulsive sound has high peak sound pressure as well as short duration time compared to the ambient noise sources. The performance of three directional estimators, the peak, RMS, and crest factor of output values, are investigated and compared with the incoherent measurement noise embedded in multiple microphone signals. The proposed formula is verified via experiments in an anechoic chamber using a uniform linear array, and the results show that the crest factor estimation of beamformer output determines the direction in a low SNR condition in which interference sources are dominants.

Contributed Paper

2:20

1pSAa4. Automating source localization and separation using sonic detection and ranging. Yazhong Lu (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, tiduslyz_01@hotmail.com)

The sonic detection and ranging (SODAR) technology developed previously (Wu and Zhu, JASA, 2013) is automated for performing blind sources localization and separation. In particular, source localization results are displayed in terms of space-frequency or space-time correlations. In other words, one can either view distributions of sound sources in three-dimensional (3-D) space versus any user-defined frequency bands, or distributions of sound sources in 3-D space versus time history. The sound pressure level (SPL) values associated with the identified sources are also calculated and displayed in these space-frequency and space-time correlation graphs. To acquire a better understanding of the distribution of sound sources together with their SPL values in 3-D space, a 3-D viewer using Google SketchUp software is employed to view these results in both space-frequency and space-time correlations. With this 3-D viewer, one is able to rotate and look at any source from any perspective, and zoom in and out to examine the details of relative positions of individual sources. The information on source locations together with windowing and filtering technologies enable us to separate the individual source signals. Examples of using this blind sources localization and separations in a non-ideal environment that involves random background noise and unspecified interfering signals are demonstrated.

MONDAY AFTERNOON, 18 MAY 2015 KINGS 3, 3:00 P.M. TO 5:00 P.M.

Session 1pSAb

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

Robert M. Koch, Chair
Chief Technology Office, Naval Undersea Warfare Center, Code 1176 Howell Street, Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Contributed Papers

3:00

1pSAb1. Prediction of broadband high-frequency acoustic reflection and transmission from subcritical membranes and elastic plates. Mauricio Villa and Donald B. Bliss (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., 148B Eng. Bldg., Durham, NC, donald.bliss@duke.edu)

Earlier work has shown that distributed broadband excitation of subcritical membranes and plates can yield surprisingly simple closed-form band-averaged directivity patterns, and that the directivity is closely related to energy flow in the structure. A first principles energy-intensity boundary element method (EIBEM) for sound fields in enclosures has also been developed, and shown to be very accurate for impedance reflection boundary conditions. The present work involves the derivation of more realistic reflection and transmission of models for enclosure boundaries with elastic boundaries. For subcritical panels, reflection, radiation, and transmission properties can be characterized by a limited set of parameters, and fairly simple analytical results are derived for the broadband case. Surface reflection and edge re-radiation are both considered. These results facilitate an energy-intensity reformulation of the structural equations, as well as the acoustic field, allowing for the possibility of formalizing the coupling between energy flows in the acoustic and structural systems. The goal is to develop efficient first principles computational models for more accurate modeling of real enclosures.

3:15

1pSAb2. Detecting building leakages using nearfield acoustic holography technique: A numerical simulation. Hirenkumar J. Patel, Kanthasamy Chelliah, Ganesh Raman (Mech. Mater. and Aerosp., Illinois Inst. of Technol., 10 W 32nd St., E1- Ste. 243, Chicago, IL 60616, hpatel58@iit.edu), Ralph T. Muehleisen, and Eric Tatara (Argonne national Lab., Lemont, IL)

A crack on a building wall and sound generated inside the building using an artificial sound source were simulated using Matlab. Various noise types affecting the pure sound signal, such as background noise in the room, were simulated. The forward problem of sound propagation from the crack surface to a virtual microphone array was simulated using the free-space green's function. Nearfield acoustic holography (NAH) was applied to perform the inverse problem to calculate the sound pressure at the crack surface and determine the crack location on the building wall. Effects of various errors and noise on the sound pressure reconstruction using NAH were studied. The reconstructed sound pressure levels on the wall surface were compared with the originally simulated sound pressure data at the crack surface. Various crack sizes and shapes were simulated to determine the correlation between the size of the crack and the reconstructed pressure field.

3:30

1pSAb3. Modal response of a simply supported plate driven by a single, asymmetric piezoelectric actuator for use as a flat-panel loudspeaker. Michael C. Heilemann and Mark F. Bocko (Elec. and Comput. Eng., Univ. of Rochester, 500 Joseph C Wilson Blvd., Rochester, NY 14627, mheilema@ur.rochester.edu)

An approximate analytical solution for the mechanical response of a thin plate to a single, asymmetrically bonded piezoelectric driver has been found. Previous work has focused only on the response of plates driven by symmetrically aligned actuators or for beams with drivers bonded to one side. The relative magnitudes of extensional and flexural waves produced by the single-sided driver configuration were determined as a function of the physical parameters of the plate and driver. The modal response of flexural waves for a simply supported thin plate also was determined. Interactions at the plate boundaries with the support structure may convert extensional waves into sound-producing flexural waves, which can lead to audio distortion. A non-moment inducing support scheme with high damping is proposed to minimize this effect.

3:45

1pSAb4. Characterization of the dominant structural vibration of hearing aid receivers. Brenno Varanda (Mech. Eng., Binghamton Univ., 1410 Alexander Way, Clearwater, FL 33756, bvarand1@gmail.com), Ron N. Miles (Mech. Eng., Binghamton Univ., Vestal, NY), and Daniel Warren (Specialty Components - Acoust., Knowles Corp., Itasca, IL)

The overall aim of this research is to analyze and characterize the mechanical vibration of hearing aid receivers, a key electro-acoustic component of hearing aids. The receiver is a high efficiency miniature sound source which utilizes a balanced armature electromagnetic motor. A standard side effect for most balance armature receivers is structural vibration. This receiver-borne structural vibration can travel through the hearing aid package to the microphones, resulting in undesirable oscillations, just like acoustic feedback. To better understand and control this important source of feedback in hearing aids, a simple dynamic model has been developed to describe the system. The model consists of two rigid bodies connected by a torsional spring and damper. A method was developed to estimate the parameters for the dynamic model using experimental data. The data were collected using translational velocity measurements using a scanning laser vibrometer of a Knowles ED-series receiver on a complaint foundation. The analytical dynamic model was validated with finite element analysis using COMSOL and the multibody dynamics module.

4:00

1pSAb5. Active and passive monitoring of valve bodies utilizing spray-on transducer technology. Kenneth R. Ledford (Acoust., Penn State, 309B EES Bldg. Penn State, University Park, PA 16802, Krl175@psu.edu), Kyle Sinding, and Bernhard Tittmann (Eng. Sci. and Mech., Penn State, University Park, PA)

Structural health monitoring (SHM) and non-destructive evaluation (NDE) can be performed both actively and passively. Active monitoring is useful for thickness measurement and crack interrogation. Passive monitoring can indicate integrity of the valve and if it is open or closed. Traditional SHM methods for valve bodies require a bonding medium that can deteriorate. A sol-gel spray-on technology eliminates the need for a coupling medium since the transducer is chemically bonded to the valve body. These spray-on transducers can be tailored to specific applications in order to maximize response or operating temperature. This technology allows for efficient on-line monitoring of valve bodies for thickness and valve integrity. The current objective is to develop a spray-on transducer and corresponding method for both active and passive SHM of valve bodies. The objective relating specifically to passive monitoring relates to indicating the condition of the valve and determining what position it is in. The active monitoring goals are to measure the thickness of the valve in critical regions, determine the presence of corrosion, and observe the roughness of the interior surface. This paper provides preliminary results on the use of spray-on transducers for on-line SHM of valve bodies for both active and passive applications.

4:15

1pSAb6. Reconstructing acoustic field based on the normal surface velocity input data. wu zhu and Sean F. Wu (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., 2100 Eng. Bldg., Detroit, MI 48202, es1699@wayne.edu)

Traditional NAH relies on the acoustic pressure collected on a hologram surface in the near field of a vibrating structure. To ensure accuracy in reconstruction, it is necessary to have a conformal array of microphones, which is not an easy task in practice. On the other hand, it is easy to use a scanning laser vibrometer to measure the normal surface velocity. This paper presents a modified Helmholtz Equation Least Squares (HELS) formulation to reconstruct the acoustic field based on the normal surface velocity input data. This approach is advantageous in that: (1) it enables one to collect input data in the far field; (2) measurement setup is very simple; and (3) the normal surface velocity contains all near-field information for reconstruction of an acoustic field. To ensure the accuracy in reconstruction, the normal surface velocity is supplemented by a few measurements of the acoustic pressure in the far field. With this combined input data, the acoustic field in three-dimensional space can be accurately reconstructed. Numerical simulations for reconstructing an acoustic field generated by arbitrarily shaped objects are demonstrated. Experimental validations on using this modified HELS formulation to reconstruct the acoustic field from a loudspeaker are also presented.

4:30

1pSAb7. Influence of ground impedance on the sound radiation of a railway sleeper. Xianying Zhang, Giacomo Squicciarini, and David J. Thompson (ISVR,Univ. of Southampton, Southampton SO171BJ, United Kingdom, xz24g12@soton.ac.uk)

A railway track consists of rails attached to sleepers (cross ties) which are laid in ballast. The sleeper provides support for the rail and transfer loads to the ballast and subgrade. Due to the wheel/rail interaction the rail is induced to vibrate and this vibration is transmitted to the sleepers; both the rail and the sleepers radiate sound. Existing models used to predict the sound radiation from the sleeper consider this to be completely embedded in a rigid ground; in reality, however, the sleeper is surrounded by, or embedded to some extent, in the ballast. It is therefore necessary to take these conditions into account in order to obtain a more realistic model. This paper investigates the influence of the ground in close proximity to the sleeper on its sound radiation. A 1/5 scale concrete sleeper is analyzed by using the boundary element method in 3-D. Ground absorption is introduced in terms of its acoustic impedance using the Delany-Bazley model and its effects on the sleeper radiation are predicted. Finally, the numerical results are validated by experimental results using a 1/5 scale model.

4:45

1pSAb8. Measurement of the moisture content of a waste paper bale based on the impact resonance test. Minho Song, Donghyun Kim, Won-Suk Ohm (Mech. Eng., Yonsei Univ., Seoul, Korea, Republic of, Yonsei University, Yonsei-Ro 50, Seodaemoon-Gu, Eng. Bldg. A391, Seoul 120-749, South Korea, songmh@yonsei.ac.kr), and Baek-Yong Um (Balance Industry Co.,Ltd., Seoul, South Korea)

The quality of a waste paper bale depends heavily on its moisture content. The higher the moisture content, the lower the quality as a recycled resource. Therefore, accurate measurement of the moisture content of a waste paper bale is a pressing concern in the paper recycling industry. In this paper, a theoretical model for a waste paper bale, an exotic and complex medium, is developed, in which the acoustic properties of the bale are assumed to be functions of the moisture content. The model is validated through a series of impact resonance tests, which show a strong correlation between the moisture content, resonance frequencies, and the associated quality factors.

Session 1pSC

Speech Communication and Psychological and Physiological Acoustics: Listening Effort II

Alexander L. Francis, Cochair
Purdue University, SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907

Christian Fullgrabe, Cochair
Institute of Hearing Research, Medical Research Council, Science Road, Nottingham NG7 2RD, United Kingdom

Chair's Introduction—1:30

Invited Papers

1:35

1pSC1. Modeling the balance between bottom-up and top-down processing in speech intelligibility tests. Verena N. Uslar, Thomas Brand, and Birger Kollmeier (Medical Phys. and Acoust., Univ. of Oldenburg, Marie-Curie-Strasse 2, Oldenburg 26129, Germany, verena.uslar@uni-oldenburg.de)

Understanding speech depends on sensory, input-driven bottom-up processing as well as on top-down processing-based, e.g., on experience—which is typically associated with more effortful listening. A qualitative and quantitative analysis of the respective contribution of both processing types is important to further diagnostics of hearing impairment, rehabilitation with hearing devices, and the predictive quality of speech intelligibility models. This talk gives a short overview of studies, which measured speech intelligibility as a function of linguistic complexity to quantify the respective contribution of both processing types. Comparison of speech reception thresholds for linguistically simple and more complex sentences in varying listening situations for younger and older adults with normal hearing and with hearing-impairment indicated that experience-driven top-down processing becomes more important with increasing listening effort. All listener groups seem to use the same mechanisms to compensate for increasing sensory or cognitive load, however the threshold for employing experience-driven top-down processing seems to lower with increasing age and hearing impairment. These results led to the development of the so called "speech processing gauge" presented here which explains the shift of the balance between bottom-up and top-down processing with regards to changes in different variables, i.e., age, hearing impairment, sensory load, and cognitive effort.

1:55

1pSC2. Effects of age, hearing loss, and linguistic complexity on listening effort as measured by working memory span. Margaret K. Pichora-Fuller (Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, Ontario L5L 1C6, Canada, k.pichora.fuller@utoronto.ca) and Sherri L. Smith (Audiologic Rehabilitation Lab., Veterans Affairs Medical Ctr., Mountain Home, TN)

Listeners, especially older or hard-of-hearing individuals, report that understanding conversation in noisy situations is effortful. Individual differences may be due to auditory and/or cognitive processing abilities. It is assumed that auditory processing depends on the individual's degree and type of hearing loss, while cognitive processing depends on the individual's working memory (WM) capacity. Many researchers have measured reading WM span to minimize the effects of hearing loss; however, variations due to hearing loss may be important in assessing processing demands (listening effort) during speech-in-noise understanding. The effects of the acoustical properties of the signal and masker on listening effort have been studied, but less is known about the effects of the linguistic complexity of the materials. We predicted that demands on WM and the correlation between WM span measures and speech-in-noise performance would increase with increasing linguistic complexity, with speech-in-noise performance correlating more strongly with auditory than with visual measures of WM span. To test these hypotheses, we administered speech tests varying in linguistic complexity and measures of both reading and listening WM span. Participants were a group of younger listeners with normal hearing and a group of older listeners with bilateral sensorineural hearing loss.

2:15

1pSC3. Task demands and cognitive abilities impact listening effort for older adult hearing aid users. Stefanie E. Kuchinsky (Ctr. for Adv. Study of Lang., Univ. of Maryland, Maryland Neuroimaging Ctr., 8077 Greenmead Dr., Bldg. #795, College Park, MD 20740, skuchins@umd.edu), Jayne B. Ahlstrom, Emily Franko-Tobin, and Judy R. Dubno (Dept. of Otolaryngol.—Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Understanding speech in background noise requires considerable effort for older adults with hearing loss, even when wearing well-fit hearing aids, but little is known about how effort varies with changes in speech recognition task demands in this population. Following research showing that effort can be indexed by changes in the autonomic pupil dilation response, the current study recorded

pupillometry as older adults wearing hearing aids completed the Words-In-Noise test (e.g., Wilson *et al*., 2007). Using a single loudspeaker at 0 degrees azimuth, participants listened to and repeated monosyllabic words in multi-talker babble at seven signal-to-noise ratios (0 to +24 dB SNR). A nonlinear relationship between word recognition score and pupil dilation was observed, with maximal effort at moderate levels of speech recognition difficulty. Individual differences in cognitive abilities (e.g., working memory, vocabulary knowledge) modulated this pattern of results. These findings highlight an important dissociation between listening effort and speech recognition task difficulty. Implications for evaluating changes in effort with speech-perception training are discussed. [Work supported, in part, by NIH/NIDCD.]

2:35

1pSC4. Speech recognition and listening effort across various conditions in adults with aphasia. Carolyn Richie (Commun. Sci. & Disord., Butler Univ., 4600 Sunset Ave., Indianapolis, IN 46205, crichie@butler.edu)

Some of the common complaints among adults with aphasia are that it is difficult to understand speech in noise and that listening can be very effortful. Nonetheless, there has been limited research to date on speech recognition in noise or listening effort for adults with aphasia. Stanfield & Richie (2014) investigated speech recognition under various listening conditions for adults with non-fluent aphasia. Participants with aphasia performed better in quiet as compared to noise, as expected, and better in noise that consisted of multi-talker babble as compared to one competing talker. They also showed modest benefit from the addition of visual cues to speech over auditory-only speech recognition. However, subjective reports of listening effort were mixed and did not line up with performance on the various tests of speech recognition. In the present study, the relationship between speech recognition under various listening conditions and an objective measure of listening effort, response time, was examined. Participants' word and sentence recognition under auditory-only and auditory-visual conditions, in quiet and in two types of noise, will be reported. The potential clinical significance to better understanding listening effort in adults with aphasia will be discussed as well.

2:55–3:10 Break

3:10

1pSC5. Using dual-task paradigms to assess listening effort in children and adults. Erin Picou and Todd A. Ricketts (Dept. of Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. S, Rm. 8310, Nashville, TN 37232, erin.picou@vanderbilt.edu)

While evidence suggests that hearing aids can reduce listening effort, mixed results are often seen when examining the effect of specific hearing aid processing. Given methodological differences across studies, we speculated that not all paradigms are equally sensitive to changes in listening effort. Moreover, many traditional paradigms may not be appropriate for children. The purpose of this project was to evaluate dual-task paradigms for measuring listening effort in adults and school-aged children. Three tasks were developed based on existing dual-task paradigms and modified to be appropriate for children. Sixteen adults (aged 22–32) and twenty-two children (aged 9–17) with normal hearing participated. All participants were tested in quiet and in noise. For all three paradigms, the primary task was monosyllable word recognition and the secondary task was a physical response time measure. The secondary tasks varied in complexity or depth of processing. Results revealed that the paradigm requiring deeper processing was most sensitive to the effects of noise for adults, but for children, the paradigm with the highest complexity was the most sensitive. Potential explanations for these group differences and the application of the most sensitive measures for future investigations will be discussed. [This work was funded by Phonak.]

3:30

1pSC6. Subjective listening effort. Michael Schulte, Melanie Krüger, Markus Meis, and Kirsten C. Wagener (Hörzentrum Oldenburg, Marie-Curie-Str. 2, Oldenburg 26129, Germany, m.schulte@hoerzentrum-oldenburg.de)

When investigating listening effort the individual subjective feedback is important to learn about and take into account the situations in which hearing impaired suffer in terms of listening effort and the relationship of listening effort to speech intelligibility. Here, we present results from a daily life questionnaire study and results from a new tool to evaluate listening effort subjectively under controlled conditions in the lab. A research questionnaire was used to determine the individual subjective listening effort in daily life in a systematic way. The questionnaire with 29 well described situations was used in a multi-center study in Denmark, Germany, and USA with 112 subjects to identify the most relevant effortful situations (e.g., "watching news in TV" or "talking at the kitchen table"). As a lab procedure we developed an adaptive categorical listening effort scaling procedure. It is based on a non-adaptive version that turned out to be a sensitive lab method in which subjects rate listening effort at predefined signal to noise ratios. The new method automatically finds the SNRs corresponding to subjective ratings from "extreme effort" to "no effort." First results show the relationship between listening effort and intelligibility for different SNRs and background noises.

Contributed Papers

3:50

1pSC7. Verbal response time in children systematically varies with the spectral resolution of speech. Kristi M. Ward, Jing Shen, Pamela E. Souza, and Tina M. Grieco-Calub (The Roxelyn & Richard Pepper Dept. of Commun. Sci. & Disord., Northwestern Univ., 2240 N. Campus Dr., Rm. 2-381, Evanston, IL 60208, kmward@u.northwestern.edu)

Children require higher spectral resolution to perform comparably to adults on speech recognition tasks. A potential source of this age-related difference is children's immature cognitive processing. Verbal response time (VRT), the speed at which listeners verbally repeat speech stimuli, has been used as a correlate of cognitive load (or listening effort, LE) in children: listeners exhibit faster VRT in conditions with greater signal-to-noise ratios and slower VRT in conditions with decreased audibility. If VRT is representative of changing cognitive demands during speech recognition, and therefore changing LE, in children, we predict that gradually decreasing the spectral resolution of the speech signal will result in a concomitant slowing of VRT. To test this prediction, we measured VRT in typically developing children (8–12 years old) while they performed a sentence recognition task that was either unprocessed (full spectral resolution) or noiseband vocoded

with 4, 6, or 8 spectral channels. Preliminary data suggest that children's VRT varies systematically with spectral resolution: VRT slows with decreasing spectral resolution of the speech signal. These results are consistent with the idea that VRT is a sensitive measure of LE in children. Discussion will consider these data in the context of other methods of measuring LE.

4:05

1pSC8. Honking is just noise (or just about): The effect of energetic masking on recognition memory for spoken words. Dorina Strori (Dept. of Psych., The Univ. of York, York YO10 5DD, United Kingdom, dorina.strori@york.ac.uk), Johannes Zaar (Tech. Univ. of Denmark, Copenhagen, Denmark), Odette Scharenborg (Radboud Univ. Nijmegen, Nijmegen, Netherlands), Martin Cooke (Univ. of the Basque Country, Vitoria, Spain), and Sven Mattys (The Univ. of York, York, United Kingdom)

Previous research indicates that listeners encode both linguistic and indexical specifications of the speech signal in memory. Recent evidence suggests that non-linguistic sounds co-occurring with spoken words are also incorporated in our lexical memory. We argue that this "sound-specificity effect" might not be due so much to a word-sound association as to the different acoustic glimpses of the words that the associated sounds create. In several recognition-memory experiments, we paired spoken words with one of two car honk sounds and varied the level of energetic masking from exposure to test. We did not observe a drop in recognition accuracy for previously heard words when the paired sound changed as long as energetic masking was controlled. However, when we manipulated the temporal overlap between words and honking to create an energetic masking contrast, accuracy dropped. The finding suggests that listeners encode irrelevant non-speech information in memory, but only in certain contexts. Calling for an expansion of the mental lexicon to include non-speech auditory information might be premature. Current work is investigating the effect in non-native listeners of English, and whether maskers that are more integral to the words and hence more difficult to segregate lead to a more robust effect.

4:20–5:05 Panel Discussion

MONDAY AFTERNOON, 18 MAY 2015 — KINGS 5, 1:00 P.M. TO 4:30 P.M.

Session 1pSP

Signal Processing in Acoustics, Architectural Acoustics, and Noise: Telecom and Audio Signal Processing

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

Invited Papers

1:00

1pSP1. Microphone cross-array beamformer processing to reduce noise and reverberation. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com), Jens Meyer (mh Acoust., Fairfax, Vermont), and Steven Backer (mh Acoust., Oakland, California)

Hands-free audio communication systems that are designed to allow audio and speech communication between remote parties are known to be sensitive to room reverberation and noise when the source is distant from the microphone. A common solution to the problem of room reverberation is to use an array of microphones to spatially filter the acoustic field. The maximum array gain is only attainable with specific microphone geometries and the gain of realizable microphone arrays is typically significantly lower than the maximum. The algorithm described in this talk attempts to address the rather slow growth in linear processing directional gain as a function of the number of microphones. One possible approach to attain higher directive gain is to use a processing scheme based on nonlinear multiplicative processing. An obvious candidate is the coherence function since it is a bounded and normalized multiplicative measure. A technique is presented for reverberation reduction processing using at least two beamforming microphones and a function of the estimated short-time coherence between the beamformer outputs. Each beamformer should have a different directional response or spatial position, or both, but with overlapping responses in the direction of the desired source.

1:20

1pSP2. Study and design of robust differential beamformers with linear microphone arrays. Jacob Benesty (INRS-EMT, Univ. of PQ, Montreal, Quebec, Canada), Jingdong Chen, Chao Pan, and Hao Zhang (Ctr. of Intelligent Acoust. and Immersive Communications, Northwestern PolyTech. Univ., 127 Youyi West Rd., Xi'an, Shaanxi 710072, China, jingdongchen@ieee.org)

Differential beamformers can generate frequency-invariant spatial responses and therefore have the great potential to solve many broadband acoustic signal processing problems such as noise reduction, signal separation, dereverberation, etc. The design of such beamformers, however, is not a trivial task. This paper is devoted to the study and design of differential beamformers with linear array geometry. The objective is to design robust differential beamformers that can form frequency-invariant beampatterns. The major contribution consists of the following aspects. (1) It discusses a general approach to the design of linear DMAs that can use any number of microphones to design a given order DMA as long as the number of microphones is at least one more than the order of the DMA. (2) It presents a method that can maximize the white noise gain with a given number of microphones and order of the DMA; so the resulting beamformer is more robust to sensor noise than the beamformer designed with the traditional DMA method. (3) It discusses how to use nonuniform geometries to further improve robustness of differential beamformers. (4) It investigates the possibility to improve the robustness of differential beamformers with the use of the diagonal loading technique.

1:40

1pSP3. Study of the maximum signal-to-noise-ratio filters for single- and multi-channel noise reduction. Jingdong Chen (INRS-EMT, Univ. of PQ, 127 Youyi West Rd., Xi'an, Shaanxi 710072, China, jingdongchen@ieee.org), Gongping Huang (Ctr. of Intelligent Acoust. and Immersive Communications, Northwestern PolyTech. Univ., Xi'an, Shaanxi, China), and Jacob Benesty (INRS-EMT, Univ. of PQ, Montreal, Quebec, Canada)

Noise reduction is a problem of recovering a speech signal of interest from its noisy observations. Since the objective of the problem is to reduce noise, thereby improving the signal-to-noise ratio (SNR), it is natural to consider the use of maximum SNR filters. However, the maximum SNR filters, if not designed properly, may introduce significant speech distortion, leading to speech quality degradation instead of improvement. This paper investigates the design of maximum SNR filters for noise reduction with minimum speech distortion. It covers the following design cases: (1) single-channel noise reduction in the time domain; (2) multichannel noise reduction in the time domain; (3) single-channel noise reduction in the short-time-Fourier-transform (STFT) domain with or without interframe information; (4) multichannel noise reduction in the STFT domain with or without interframe information. A large number of experiments are performed to illustrate the properties of these maximum SNR filters.

2:00

1pSP4. Coded excitation in space and time for the identification of geometric parameters using audio transducer arrays. Jung-Woo Choi (Mech. Eng., KAIST, 291 Daehak-ro, Yuseong-gu, Daejeon 305-701, South Korea, jwoo@kaist.ac.kr)

Impulse responses of a room can provide a lot of geometric information of acoustic systems. Loudspeaker and microphone array positions, locations of walls and scattering bodies are representative examples. To extract geometric information from the room impulse responses, the measurement time should be short enough such that the degradation of time delay estimation performance due to non-stationary environmental noises can be minimized. In this work, we attempt to enhance the SNR of the short-term measurement, by using a coded excitation technique in space and time. Both the spatially and temporally coded excitation signals are applied to a set of loudspeakers combined with microphones, and it is shown that the measurement time can be shortened with minimal loss of SNR and time delay estimation performance.

2:20

1pSP5. Adaptive beamforming for acoustic echo cancellation using measured array models and subband filtering. Mingsian R. Bai and Li-Wen Chi (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan 30013, Taiwan, msbai63@gmail.com)

Acoustic echo that can substantially undermine speech quality is one of the key issues must be addressed in practical telecommunication systems. In this paper, an evolutionary exposition is given in regard to the enhancing strategies for acoustic echo cancelers (AEC). On the basis basic adaptive filtering, a highly directional microphone array is designed as a fixed beamformer (FBF) to focus on the near-end speaker and suppress the echo from the far-end as well as noise and interference from the background. Design of the beamformer is based on array models interpolated from measured frequency responses. Subband (SB) filtering with polyphase decomposition is exploited to accelerate the cancellation process with multiple choice of step size in the subbands. To further enhance the canceler, adaptive generalized sidelobe canceler (GSC) can be utilized, which is comprised of a FBF for the near-end speech and an adaptive blocking module for the echo and noise. Objective tests in terms of echo return loss enhancement (ERLE) and perceptual evaluation of speech quality (PESQ) are conducted to compare four algorithms with various combinations of enhancement. The results show that the GSC-SB-AEC approach has attained the highest ERLE and best speech quality, even in a double-talk scenario.

2:40–3:00 Break

Contributed Papers

3:00

1pSP6. Experimental verification of a Fisher Information Model for azimuth estimation in broadband active sonar. Colin J. Ryan (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 679 Main St., Harwich, MA 02645, cryan2@umassd.edu), John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, North Dartmouth, MA), and Laura N. Kloepper (Biology, Brown Univ., Providence, RI)

Bats easily identify targets of interest and reject clutter in the wild. A bat's transmit and receive beampatterns are frequency dependent, so it is possible that they exploit spectral cues to distinguish on-axis targets from off-axis clutter. The frequency dependent beampattern can also be interpreted as a different lowpass filter for each angular arrival. Given the known transmitted signal, the angle of arrival can be estimated from the high frequency attenuation in the received signal. The broadband Fisher information (FI) for azimuth in active sonar predicts a maximum FI at a slightly off-axis location, suggesting bats might target their sonar beam askance of a target to maximize the precision in estimating azimuth. This paper tests the FI model in a laboratory with a baffled tweeter as the sonar transmitter and an omnidirectional receiver. The experimentally measured beampatterns are used in Monte Carlo simulations with a maximum-likelihood estimator for azimuth over a range of SNR. The experimental results gained from the laboratory tests are compared with the model to assess the agreement between the predicted azimuth of maximum FI and the observed azimuth minimizing experimental error variance. [Funded by UMass Dartmouth Office of Undergraduate Research, College of Engineering, and ONR.]

3:15

1pSP7. Developing an audio network system for teleconferencing with evaluation of audio quality. Hansol Lim, Hyung Suk Jang, Muhammad Imran, and Jin Yong Jeon (Architectural Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, sollim0128@gmail.com)

A teleconference system was developed to transmit speech signals for audio communication with remote users. The hardware system was composed of 3-D microphone arrays to capture directional sound and a wireless headset to provide freedom of movement for users. An application programming interface (API) was used for interaction between the computer and the audio devices, including the microphone, headphone, and ADDA converter, transferring input/output audio signals in real time through the network. For the evaluation of the system, a latency test was performed with a simultaneous recording system in capturing and reproduction positions. The latency from the network procedure was calculated with different buffer sizes, and the lower-delay and smaller-deviation buffer was selected. In addition, the transmitted audio signals were compared with the input signals in terms of signal patterns and frequency responses. The resulting audio qualities in the time and frequency domain were suitable for a teleconferencing system.

3:30

1pSP8. Tone-to-text applications for sub-Saharan African tone languages. Aaron Carter-Enyi (The Ohio State Univ., 2180 N 4th St., Columbus, OH 43201, cartercohn@gmail.com)

Based on experimental findings, this paper proposes a framework for computer recognition of speech tones in Niger-Congo languages. This language family includes over 1000 languages in Sub-Saharan Africa and approximately one billion speakers. Many of these languages use pitch contrast to differentiate words in a system of two or more pitch levels (e.g., high, mid, and low). The results of two new studies conducted in Nigeria from 2013-14 indicate speech-tone is perceived as tonemic intervals (e.g., high-high, high-mid, and high-low). An experimental study (n = 1448) identified ranges of pitch difference that form perceptual categories for these tonemic intervals. An initial application of these findings is adding tone diacritics to text by interpreting fundamental frequency. Representing tone in text has been a persistent problem for Niger-Congo languages and smartphones are ill-equipped for marking tone diacritics. A signal processing application that recognizes speech tones and marks them on computer text would be more efficient than manually highlighting and marking each syllable. Frequency distribution of tones is dynamic between speakers. Thus, there are potential benefits of using machine learning to create speaker-dependent software. Because the proposed method relies on existing algorithms for fundamental frequency, the problem of estimation errors will also be addressed.

3:45

1pSP9. Subarray based localization using extended coprime arrays. Andrew T. Pyzdek, R. Lee Culver, and Dave Swanson (Appl. Res. Lab., The Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu)

Partitioning of long linear arrays into a number of smaller subsections, termed subarrays, is a common form of processing used both to compensate for irregularity in array shape and to localize near-field sources. While this manner of processing is applicable to uniform linear arrays, as the subarrays are similarly uniform and possess the same minimum element spacing, subarray methods cannot always be applied to sparse arrays with uneven spacing along their length. One sparse array design of interest is the extended coprime array, an array composed of two uniformly spaced component arrays, each undersampled by integer factors which are selected to be coprime. By exploiting the regularity of spatial lag repetitions in extended coprime arrays, we show that appropriately selected subsections of a coprime array can be used as subarrays to determine distance to a near-field source. The performance of the coprime array processed in this manner will be compared to similar processing performed on alternative sparse array designs and the baseline performance of a uniform linear array of equal aperture.

4:00

1pSP10. Testing spatial co-prime sampling theory. Radienxe Bautista and John R Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, rbautista@umassd.edu)

An array of sensors can be used to estimate the direction of arrival of a narrowband signal in the far field with the use of conventional beamforming. In order to avoid spatial aliasing, the distance between the sensors have to be $d \leq \lambda/2$ apart. This is analogous to the Nyquist theorem for sampling in time with $f_s = 2f_o$. Co-Prime arrays are non-uniform arrays that can predict what a uniform array with an aperture of $L = M \times N$ total sensors can using fewer sensors; they have a total number of sensors $L = M + N—1$ while maintaining an aperture of $M \times N$ sensors in a full uniform line array with $\lambda/2$ spacing between elements. Each subarray is uniform linear, equally spaced by $N\lambda/2$ or $M\lambda/2$ and then combined to create a non-uniform linear array. The result with the different inter-element spacing between the subarrays is grating lobes in different locations with the exception of the true DOA. Conventional beamforming is done with both subarrays and the outputs are then multiplied together to yield a single beam pattern. The theory to this will be tested using a uniform array of 30 sensors and selecting the data from certain sensors to achieve co-prime sampling.

4:15

1pSP11. The study of adaptive time reversal mirror as a crosstalk mechanism for underwater communication. Yi-Wei Lin (Systems and Naval Mechatronics Eng., National Cheng Kung Univ., No.1, University Rd., Tainan 701, Taiwan, ls5028@gmail.com)

The time reversal mirror technique has been widely applied to mitigate the inter-symbol interference in underwater channels. Meanwhile, an adaptive time reversal mirror is introduced to improve the crosstalk quality between receivers in underwater communication. To explore the effectiveness of this method, this study extended the analysis of adaptive time reversal mirror as a crosstalk mechanism and explored the mechanism in an experiment using a towing tank as a testing platform. The advantage of this process is its simplicity in examining the effects of the array configuration of this crosstalk mechanism. Results of parametric experiments are discussed i.e. the effects of number of receivers, spacing between sources and noise energy threshold. Experimental data at 10 and 16 kHz with a 5-kHz bandwidth demonstrate as much as an 8-dB signal to noise ratio improvement for four receivers dual sources over a 30 m communication range in a 3.3 m depth testing platform. The results indicate array configuration affects this mechanism drastically.

Session 1pUW

Underwater Acoustics: Environmental Characterization, Localization, and Vectors

David J. Zartman, Cochair
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Jeffery D. Tippmann, Cochair
Marine Physical Laboratory, Scripps Institution of Oceanography, 9500 Gilman Drive, La Jolla, CA 92093

Contributed Papers

1:00

1pUW1. Array shape calibration using long-term ambient noise records. Stephen Nichols and David L. Bradley (Appl. Res. Lab., The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, smn5198@psu.edu)

When using an acoustic array to determine the bearing of a source, errors in the sensor positions severely reduce the accuracy of any bearing measurements. This issue is particularly problematic for three sensor arrays, designed to work in two dimensions, because of the lack of redundancy built into the array. This paper presents a method for correcting errors in sensor positions. It is mathematically straightforward to determine the bearing of a far-field impulsive signal by determining the time difference of arrival between sensors by cross-correlation. These time delays can also be used to estimate the sound speed in the vicinity of the array. In an isotropic acoustic medium, the local sound speed is expected to be independent of the source bearing. If the sensor positions used to determine the sound speed and bearing are incorrect, the resulting sound speed measurements will be bearing-dependent. Using an analytically derived function, the correct array shape (with only translational and rotational ambiguity) can be backed out from the directional sound speed distribution. This method will be demonstrated using long-term ambient noise records from the Comprehensive Nuclear-Test Ban Treaty Organization's (CTBTO) hydrophone arrays.

1:15

1pUW2. Maximum entropy inference of seabed attenuation parameters using ship radiatedbroadband noise. David P. Knobles (ARL, Univ. of Texas, PO Box 8029, Austin, TX 78713, knobles@arlut.utexas.edu)

The use of ambient noise as a means of inferring physical properties of the ocean is of significant interest in ocean acoustics. This study employs the received acoustic field generated by the passage of the R/V Knorr on the New Jersey continental shelf to test a new approach that estimates the probability distributions of both the aspect dependent source levels and the parameters that represent the geoacoustic structure of the seabed. Since the source levels and the environmental parameters have an intrinsic ambiguity, both classes of parameters must be included in the model hypothesis space as random variables. The statistics of the error function, needed to uniquely specify the likelihood function, are estimated with a maximum entropy approach by creating a data ensemble that includes samples from time periods where the ship-receiver geometry is dominated by stern, bow, and port aspect. This method has its origins from the observation that Bayes' rule is symmetric with the interchange of the data space and the hypothesis space. [Research supported by ONR 32 OA.]

1:30

1pUW3. High frequency source localization in a shallow ocean sound channel using frequency-difference matched field processing. Brian M. Worthmann and David R. Dowling (Univ. of Michigan, 2010 Lay Automotive Lab, Ann Arbor, MI 48109, bworthma@umich.edu)

Matched field processing (MFP) is an established technique for locating remote acoustic sources in known environments. Unfortunately, environment-to-propagation model mismatch prevents successful application of MFP in many circumstances, especially those involving high frequency signals. For beamforming applications, this problem was found to be mitigated through the use of a nonlinear array-signal-processing technique called frequency difference beamforming (*Abadi et. al*. 2012). Building on that work, this nonlinear technique was extended to Bartlett MFP, where ambiguity surfaces were calculated at frequencies two orders of magnitude lower than the propagated signal, where the detrimental effects of environmental mismatch are much reduced. Previous work determined that this technique has the ability to localize high-frequency broadband sources in a shallow ocean environment with a sparse vertical array, using both simulated and experimental propagation data. Using simulations, the performance of this technique with horizontal arrays and adaptive signal processing techniques was investigated. Results for signals with frequencies from 10 kHz to 30 kHz that propagated in a 100-m-deep shallow ocean sound channel with a downward refracting sound speed profile will be shown for source array ranges of one to several kilometers. [Sponsored by the Office of Naval Research.]

1:45

1pUW4. Passive acoustic source localization using sources of opportunity. Christopher Verlinden, Jit Sarkar, William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0701, cmverlin@ucsd.edu), Karim Sabra (Mech. Eng. Dept., Georgia Inst. of Technol., Atlanta, GA), and William Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Rather than use replica fields for matched field processing (MFP) derived from acoustic models requiring detailed environmental input, we demonstrate that data derived replicas from ships of opportunity can be used for assembling a library of replicas for MFP. The Automatic Identification System (AIS) is used to provide the library coordinates for the replica library and a correlation based processing procedure is used to overcome the impediment that the replica library is constructed from sources with different spectra and will be used to locate another source with it own unique spectral structure. The method is illustrated with simulation and then verified using acoustic data from a 2009 experiment for which AIS information was retrieved from the Unites States Coast Guard Navigation Center (USCG NAVCEN) Nationwide AIS (NAIS) database.

2:00

1pUW5. Bayesian linearized two-hydrophone localization of a pulsed acoustic source. Emmanuel Skarsoulis (Foundation for Res. and Technol. - Hellas, Heraklion, Greece) and Stan E. Dosso (Univ. of Victoria, University of Victoria, Victoria, British Columbia, Canada, sdosso@uvic.ca)

A three-dimensional localization method for transient acoustic sources is developed, based on time differences between direct and surface-reflected arrivals at two hydrophones. The method accounts for refraction caused by a depth-dependent sound-speed profile using a ray-theoretic approach, and, further, it provides localization error estimates accounting for uncertainties of the arrival times and hydrophone locations, as well as for depth-dependent uncertainties in the sound-speed profile. In the first of two steps, source depth and range to each hydrophone are estimated using an iterative, linearized Gauss-Markov inversion scheme. In the second step, the estimated source ranges are combined with the hydrophone locations to obtain the source location in the horizontal. Localization performance is analyzed in a simulation study, and the linearized localization estimates and uncertainties are validated by comparison with a fully-nonlinear (but numerically intensive) Markov-chain Monte Carlo inversion. [Work supported by Aristeia-II program, EU-ESF and Greece, NSRF 2007-13.]

2:15

1pUW6. Sound speed structure monitoring of the Antarctic Ocean by new style float. Shinpei Gotoh (Intelligent Interaction Technologies, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8573, Japan, gotohs@jamstec.go.jp), Toshio Tsuchiya, Yoshihisa Hiyoshi (JAMSTEC, Yokosuka, Kanagawa, Japan), and Koichi Mizutani (Intelligent Interaction Technologies, Univ. of Tsukuba, Tsukuba, Japan)

JAMSTEC developed the new profiling float "Deep NINJA" for deep-sea type, was subjected to long-term monitoring of one year in the Antarctic Ocean off the coast of Adelie Coast from 2012. As a result, succeeded in the long-term monitoring of the sound speed profile to depth of the 4000 m in the Antarctic Ocean for the first time in the world, and was able to capture a seasonal change in the surface area in the freezing season and the thawing season. In addition, calculating a sound speed from these data, simulations were performed assuming the low frequency sonar. The result was obtained, the ingredient which propagates while repeating a reflection in the extremely small layer of the sea surface neighborhood, and the ingredient that propagates while being reflected in near the water depth 100m which changes of sound speed gradient. From this, propagation loss was small in winter than summer, and was shown a possibility that a sound wave would propagate to a more distant place. This may affect the long-distance sound wave propagation of the echo locations of the passive sonar and marine mammals.

2:30

1pUW7. A deep-water experiment to measure mid frequency attenuation. Jeffery D. Tippmann, Jit Sarkar (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, jtippman@ucsd.edu), Philippe Roux (Institut des Sci. de la Terre, Universite Joseph Fourier, CNRS UMR 5275, Grenoble, France), William S. Hodgkiss, and William A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

A deep-water experiment was performed off the west coast of the United States with a short vertical array cut for 7.5 Hz and a source transmitting tonals as well as chirps. The data were processed to identify eigenray arrivals out to a convergence zone for the ultimate purpose of revisiting mid frequency, deep water volume attenuation in the Pacific Ocean and comparing it to decades old measurements and models. Further, aside from the beamforming done at the receiver array, attempts were made to construct a synthetic aperture source array in order to perform "double beamforming" and thereby enhance the signal to noise ratio. We present our initial results of this experiment.

2:45

1pUW8. Symmetric and asymmetric reversible quasi-holographic processing of dual transducer sonar for feature extraction and analysis. David J. Zartman, Daniel S. Plotnick, and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Phys. and Astronomy Dept., Pullman, WA 99164-2814, zartman.david@gmail.com)

Two independent transducers in a side-by-side configuration may transmit and/or receive in-phase or out-of-phase with each other. In the transmit mode, the transducers are driven individually; in the receive mode the transducers are recorded independently, allowing the phase relationships to be determined in post-processing. This results in both monostatic and weakly bistatic data. When the transducers are scanned together, as for synthetic aperture sonar, the scattering results from glory target features of a small solid sphere in water are seen to be sensitive to different combinations of in-phase and out-of-phase sources/receivers. Certain combinations function as a spatial derivative and may be generated synthetically from scans using a single transducer. This spatial derivative is shown to be highly sensitive to the alignment of a solid cylinder in water. Imaging via reversible quasi-holographic processing [D. J. Zartman, D. S. Plotnick, T. M. Marston, and P. L. Marston, Proc. Meet. Acoust. **19**, 055011 (2013)] was applied to this target and used for target isolation, feature extraction, identifying and separating system effects from target physics, and for self-normalization. Though transmitting and receiving in-phase improves ordinary SNR, it can reduce some feature sensitivity relative to differential processing. [Work supported by ONR.]

3:00–3:15 Break

3:15

1pUW9. Geo-acoustic inversion of marine sediments using vector sensor measurements. David Dall'Osto (Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@u.washington.edu) and Peter H. Dahl (Appl. Phys. Lab. and Dept. of Mech. Eng., Univ. of Washington, Seattle, WA)

In this paper, a narrow-band geo-acoustic inversion scheme based on the acoustic vector field is presented. Measurements of acoustic particle velocity made with a vector sensor and short line array of hydrophones are studied with regards to the interference pattern generated by single 1–4 kHz source. This source was lowered from a research vessel positioned within three water depths (depth 20-m) of the receivers, and projected a series of 100-ms long continuous wave (cw) pulses as it was raised toward the surface. The interference pattern measured at the receivers is assessed by a non-dimensional property of the acoustic particle velocity, the degree of circularity [Dall'Osto and Dahl, J. Acoust. Soc. Am. **134**, 109 (2013)], computed directly at the vector sensor and approximated along the line-array. This degree of circularity is highly sensitive to the phase of interfering multipaths, which depends on both the source-receiver geometry and the geo-acoustic properties of sea-bed. Source positions measured by GPS and depth sensors are used to model the time-dependent particle velocity field as the cw tone establishes a steady-state response. A best-fit model to the field data provides an estimate of the geo-acoustic properties of the sandy sediment at the experimental site.

3:30

1pUW10. Localization for broadband source by a single vector sensor in shallow water. Junjie Shi, Dajun Sun, Yunfei Lv (Sci. and Technol. on Underwater Acoust. Lab. and College of Underwater Acoust. Eng., Harbin Eng. Univ.,Rm. 1005 of Shuisheng Bldg. of Harbin Eng. University, Harbin 150001, China, junjieshi@hrbeu.edu.cn), and Yun Yu (Naval Acad. of Armament, 100161, Beijing, China)

The physical meaning of array invariant is the absolute traveling time of different modal from a source to a receiver under the average sound speed. It can be gotten by the derivative of the arriving time against the corresponding cosecant function of elevation angle for each modal, which can be used for the passive source localization. A vector sensor can be seen as a small volumetric array. It has its capability of elevation angle determination to describe the multi-modal propagation and the single-modal dispersion in shallow water, which may imply a chance to get the array invariant for the source localization. Here, we take advantage of the dispersion based short time Fourier transform and warping transform techniques for the time-frequency analysis of vector sensor signals. This technique can improve acoustic normal mode identification and thereby extract the modal data more accurately for array invariant determination. The idea had been effectively validated during the experiment that took place in October 2014 in South China Sea of nearly 100 m depth. [Work supported by the National 863 Project (No. 2011AA090502) and National Natural Science Funds Program (No. 11404406).]

3:45

1pUW11. Study on the interference structure of vector acoustic field in shallow water. Dajun Sun, Junjie Shi, Jidan Mei (Sci. and Technol. on Underwater Acoust. Lab. and College of Underwater Acoust. Eng., Harbin Eng. Univ.,Shuisheng Bldg. of Harbin Eng. University, Harbin, China, junjieshi@hrbeu.edu.cn), and Haizhu Xu (Naval Acad. of Armament, 100161, Beijing, China)

The application of vector sensors is closely associated with the characteristics of vector acoustic field in shallow water, whose study is the foundation for target detection, positioning, and classification. Beginning from Normal Mode theory, theoretical studies are presented about the amplitude of different components of vector acoustic field involving the pressure, horizontal and vertical particle velocity, as well as the phase relation between the pressure and the vertical particle velocity. The main factors considered here are the source depth and the range relative to the vector sensor. These interference structures were observed during the experiment that took place in July of 2013 in South China Sea of nearly 90m depth. The results indicated that the interference structures of vector acoustic field are not only influenced by the sound speed profile but also by seabed parameters including stratification depth and its corresponding acoustic properties. Moreover, the phase relation between the pressure and vertical particle velocity is remarkably impact by these parameters. [Work supported by the National 863 Project (No. 2011AA090502) and Defense Industrial Technology Development Program (B2420132004).]

4:00

1pUW12. Second-order statistics of the instantaneous mutual information in time-varying underwater acoustic particle velocity channels. Chen Chen, Shuangquan Wang (Samsung Mobile Solution Lab, San Diego, CA), and Ali Abdi (New Jersey Inst. of Technol., 323 King Blvd., Newark, NJ 07102, ali.abdi@njit.edu)

Instantaneous mutual information (IMI) is the amount of information that a time-varying channel can convey for the given time instant. In this paper, second-order statistics of IMI are studied in time-varying underwater acoustic particle velocity channels. First, the autocorrelation function, correlation coefficient, level crossing rate, and average outage duration of IMI are provided in a time-varying fading channel. Exact expressions are given in terms of the summation of special functions, which facilitate numerical calculations. Then, accurate approximations for the autocorrelation function and correlation coefficient are presented for low and high signal-to-noise ratios. Moreover, analytical and numerical results are provided for the correlation and level-crossing characteristics of IMI in underwater particle velocity channels. The results shed light on the dynamic behavior of mutual information in underwater acoustic particle velocity channels. [Work supported in part by the National Science Foundation (NSF), Grant CCF-0830190.]

4:15

1pUW13. Comparative sensitivity of pressure gradient receivers of force and inertial types to sound pressure in plane wave. Vladimir Korenbaum (Pacific Oceanologic Inst., 43, Baltiiskaya Str., Vladivostok 690041, Russian Federation, v-kor@poi.dvo.ru), Sergei Gorovoy (Far Eastern State Univ., Vladivostok, Russian Federation), Alexandr Tagiltcev, and Anatoly Kostiv (Pacific Oceanologic Inst., Vladivostok, Russian Federation)

Among known constructing schemes of pressure gradient receivers (PGR)—a difference type (including two-point version), a force type, and an inertial one—only the last two types are applicable when receiver size is much smaller than a wavelength of plane wave. Each of these two PGR types has certain advantages/disadvantages for various applications. The objective is a comparison of these PGRs, having identical external dimensions, by their sensitivity to sound pressure. In the long-wave two-dimensional approach the expression for a sensitivity to sound pressure is derived for a force type PGR, in which plate bending piezoelectric transducer is installed in a passage of the cylindrical housing. Also, the expression is developed for the sensitivity to sound pressure of an inertial type PGR with a one-component accelerometer embedded into its cylindrical housing. We derived the expression to estimate required one-side pressure sensitivity of the bending piezoelectric transducer of the force type PGR to equalize the sensitivities of PGRs of both types to sound pressure when the vibration sensitivity of the accelerometer is known. Possibilities of increasing sensitivity of the force type PGR are considered. [The study was supported by the grant 15-IV-1-001 of Far Eastern Branch of Russian Academy of Sciences.]

4:30

1pUW14. A wave structure based method for recognition of marine acoustic target signals. Qingxin Meng and Shie Yang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St.,Nangang District, Harbin City, Heilongjiang Province 150001, China, mengqingxin005@hrbeu.edu.cn)

The loudness and timbre of propeller are remarkable features of ship-radiated noise. The information of loudness and timbre is indicated in the wave structure of time series, and the feature of wave structure can be applied to classify various marine acoustic targets. In this paper, the feature extracting method of time series wave structure is studied. A nine-dimensional feature vector is constructed through statistical characteristics, containing zero-crossing wavelength, peek-to-peek amplitude, zero-crossing-wavelength difference and wave train areas. The feature vectors are inputted into SVM classifier to identify marine acoustic targets. The kernel function is set radial basis function (RBF).The penalty factor and kernel width of RBF are selected by the method of grid search. Finally, the recognition rate of test data reaches over 89.5%, with the help of cross validation. The sea-test data show the validity of target recognition ability of the method above.

4:45

1pUW15. Possibility of acoustic noise interferometry applications for passive remote sensing in shallow water. Sergei Sergeev, Andrey Shurup, Alisa Scherbina, and Pavel Mukhanov (Phys., Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, sergeev@aesc.msu.ru)

The cross-coherence function of the ambient noise field received during the quite long accumulation time by two hydrophones in the shallow water can exhibit two peaks. These peaks correspond to the travel times of signals between these two hydrophones. If the features of the shallow water are changed than the positions of these peaks are shifted. As result one can get information to reconstruct the physical properties and variability of the underwater environment using such peaks' shifts. In this work, the possibility of the passive remote sensing in shallow water features is considered based on the time-frequency analysis of the ambient noise cross-coherence function. The key point which define the possibilities of this method are the accumulation times required to estimate the peaks of the noise cross-coherence function with the acceptable signal-to-noise ratio. It is shown that the desired accumulation times can be considerably reduced by the appropriate choice of the frequency bands where the noise fields are formed by the small number of the incoherent hydroacoustic modes.

5:00

1pUW16. Eigenvector-based signal subspace estimation. Jorge E. Quijano (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd. (Ring Road), Victoria, British Columbia V8P 5C2, Canada, jorgeq@uvic.ca) and Lisa M. Zurk (Elec. and Comput. Eng. Dept., Portland State Univ., Portland, OR)

In this work, we explore the performance of a new algorithm for the estimation of signal and noise subspaces from limited data collected by a large-aperture sonar array. Based on statistical properties of scalar products between deterministic and complex random vectors, the proposed algorithm

defines a statistically justified threshold to identify target-related features (i.e., wavefronts) embedded in the sample eigenvectors. This leads to an improved estimator for the signal-bearing eigenspace that can be applied to known eigenspace beamforming processors. It is shown that data projection into the improved subspace allows better detection of closely spaced targets compared to current subspace beamformers, which utilize a subset of the unaltered sample eigenvectors for subspace estimation. In addition, the proposed threshold gives the user control over the maximum number of false detections by the beamformer. Simulated data are used to quantify the performance of the signal subspace estimator according to a normalized metric that compares estimated and true signal subspaces. Improvement on beamforming resolution using the proposed method is illustrated with simulated data corresponding to a horizontal line array, as well as experimental data from the Shallow Water Array Performance experiment.

5:15

1pUW17. Seismic sources in parabolic equation solutions for beach and island propagation scenarios. Scott D. Frank (Mathematics, Marist College, 3399 North Rd., Marist College, Poughkeepsie, NY 12601, scott.frank@marist.edu) and Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, Golden, CO)

Seismic events at and under the seafloor can be detected by seismometers on a beach. Events occurring in the earth's crust without ocean cover can also be detected at sea. Parabolic equation solutions for environments that include seismic sources in conjunction with island topography demonstrate that results for propagation moving away from or onto land can be obtained. Transmission loss results are given in the context of the (u_r,w) parabolic equation formulation of elasticity. Horizontal and vertical displacement values, as would be measured on land-based or ocean bottom seismometers, can also be obtained. Examples involving underwater volcanic events are considered, as are shore-bound events that generate oceanic *T*-waves that would be measured at sea.

MONDAY AFTERNOON, 18 MAY 2015 DUQUESNE 1, 5:00 P.M. TO 6:00 P.M.

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

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

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

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 108, Mechanical vibration, shock, and condition monitoring, and four of its subcommittees, take note-that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 19 May 2015.

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

Session 1eED

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

Cameron T. Vongsawad, Cochair
Physics & Astronomy, Brigham Young University, 1041 E. Briar Avenue, Provo, UT 84604

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

This workshop consists of a hands-on tutorial, interactive demonstrations, and a panel discussion about careers in acoustics. The primary goals of this workshop are to expose kids to opportunities in science and engineering and to interact with professionals in many areas of acoustics. A large number of volunteers are needed to make this a success. Please email Traci Neilsen (tnb@byu.edu) if you have time to help with either guiding the kids to the event and helping them get started (5:00 p.m. to 6:00 p.m.) or exploring principles and applications of acoustics with small groups (5:00 p.m. to 7:30 p.m.). We will provide many demonstrations, but feel free to contact us if you would like to bring your own.

Contributed Paper

5:30

1eED1. Visualization of multiple array acoustic source localization. Michael V. Scanlon (US Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, michael.v.scanlon2.civ@mail.mil)

Transient acoustic events such as gunfire, rocket launches, and explosions can be located using triangulation with multiple lines-of-bearing (LOBs) generated by distributed acoustic microphone arrays. The Army has fielded diverse sound localization and ranging systems that work well in open field environments, but suffer reduced performance in urban environments where echoes, multipath, diffraction, and blockages modify both the signatures and the propagation path. This demonstration will use light projected through a water basin to a screen to show traveling waves (ripple diffraction patterns) from transient events and how those wavefronts pass differently across the distributed array locations. The observers will gain an understanding of how multiple microphone elements in a small cluster of microphones (an array) can derive time-difference-of-arrival (TDOA) estimates to create array LOBs, as well as an introduction to triangulation through multiple distributed array LOBs. Objects simulating walls and buildings will be placed in the basin to demonstrate how complex waveforms and propagation paths can decrease the accuracy of acoustic localization systems.

Payment of separate registration fee required to attend.

MONDAY AFTERNOON, 18 MAY 2015 BALLROOM 3, 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Man-Made Noise and Aquatic Life: Data, Data Gaps, and Speculation

Micheal L. Dent, Chair
Psychology, University at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260

Invited Paper

7:00

1eID1. Man-made noise and aquatic Life: data, data gaps, and speculation. Arthur N. Popper (Dept. of Biology, Univ. of Maryland, 15501 Prince Frederick Way, Silver Spring, MD 20906, apopper@umd.edu)

This talk will consider how man-made sounds may impact aquatic life—with a focus on fishes (with possible digressions to invertebrates, marine mammals, and turtles). The talk will start with a discussion of a Jacques Cousteau movie and then ask why animals (and humans) hear. After considering how fishes (and invertebrates) hear, the focus of the talk will turn to potential impacts of sound on these animals and what is currently actually known (based on data and not speculation) about effects on physiology and behavior. The talk will conclude with a consideration of recent guidelines on effects of sounds and then provide an overview of data gaps and the most important areas for future research.