Session 2aAAa

Architectural Acoustics and Noise: Forensic Studies from Noise Identification to Solutions in the Built Environment

Kenneth Cunefare, Cochair
Georgia Tech, Mechanical Engineering, Atlanta, GA 30332-0405

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

Chair's Introduction—8:00

Invited Papers

8:05

2aAAa1. What's that sound? Probe tones and poltergeists: Two case studies in diagnosing noise sources. Kenneth A. Cunefare (Georgia Tech/Arpeggio Acoust. Consulting, Mech. Eng., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

Case #1: Following construction of a new hospital building within a larger campus, residents began to complain about droning noises in the surrounding neighborhood. Blame was laid on HVAC chillers and an MRI chiller installed on exposed roof-tops of the building. However, there were numerous other chillers on buildings on the campus that were not blamed; a test using probe tones eliminated the hospital building as the source impacting the community. Case #2: A homeowner complained of intermittent groaning and moaning noises in the home (poltergeist-like). Interviews revealed that the noises occurred essentially randomly; while the homeowner correlated occurrences to whether or not it had recently rained, the correlation was weak. Additionally, the homeowner had found that opening a water tap would cause the sound to fade away; this was suggestive, but since the sound faded away in a short time even without opening the valve, it was not definitive. Complicating the diagnosis was the intermittency of the noise (weeks could pass between events), and its short duration. A faulty GFI threw a red-herring into the mix, but ultimately the problem was solved by replacing the household water pressure regulator.

8:25

2aAAa2. Trains, transformers, and tinnitus: Case studies in hard-to-detect sounds. Arno S. Bommer and Edgar Olvera (CSTI Acoust., 16155 Park Row, Ste. 150, Houston, TX 77084, arno@cstiacoustics.com)

It's a phone call many acoustical consultants dread: someone driven crazy by an annoying sound. Sometimes, the source is unknown. Sometimes it's a nearby industry or antagonistic neighbor. Some sounds are intermittent; some are steady. Some are high frequency; others are low and are described as vibration. Most consultants are greatly relieved if they can hear and measure the sound themselves. But the source can still be elusive, especially if the sound is very faint and if there are multiple potential sources. More often than not, the consultant can hear and measure nothing, which gets blamed on his poor hearing and shoddy equipment. A discussion of tinnitus is often received as an accusation of "hearing things." But is tinnitus (or a similar condition) the only possible explanation? Why is the sound heard only at home? What triggered the person to start hearing it? The person may be especially sensitive, but to what? What advice is useful? The authors will present several case histories where community sounds were ultimately identified. For those sounds not identified, the author will discuss a number of possible explanations along with recommendations for the consultant and the person hearing the sound.

8:45

2aAAa3. The singing parking garage: Wind induced Helmholtz resonation of perforated metal panels. Erik Miller-Klein (SSA Acoust., LLP, 222 Etruria St., Ste. 100, Seattle, WA 98109, erik@ssaacoustics.com)

An above ground parking garage with perforated metal facade experienced high amplitude sound when northwest winds exceeded 10 mph around this building. The noise only occurred under very specific weather conditions, which required on-call noise and vibration measurements to identify the cause and feasible solution. The noise was traced to the Helmholtz resonator effect caused by the perforated metal panel holes and the air cavity associated with the headlight screens behind these perforated facade panels. Through our on-site investigation and analysis, the root cause was identified and a low cost remediation plan was designed and implemented.

9:05

2aAAa4. Methods for the identification of intermittent and unpredictable noises. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

Acoustical consultants are often asked to identify noises that are intermittent and unpredictable, such that attended observation is impractical. This paper presents methods for recording and identifying such events using unattended long-term recordings of sound and vibration, and for efficiently analyzing the voluminous data that can accumulate during such studies.

9:25

2aAAa5. Room acoustic design, measurement, and simulation techniques to reduce hospital noises within patients' environment. Mojtaba Navvab (Architecture, Univ. of Michigan, 2000 Bonisteel Blvd., Art and Architecture Bldg., Ann Arbor, MI 48109-2069, moji@umich.edu)

Hospitalized patients and clinical staff identify noise as a major stressor. Environmental hospital noise raises ambient noise levels significantly above ideal levels. Options to reduce hospital noise include methods to modify noise, such as closing doors, adjusting hospital equipment, hospital personnel behaviors, and clinical alarms such as wireless communication devices for staff and future "smart" algorithms for patient-specific alarm thresholds. This study utilizes the beamforming techniques to localize and examines basic characteristic of sound field through measurements and simulation of noise in an environment that affects the patient's audio perception, and collects evidence for impact of noise intensity on patient hearing toward reduction of the noise sources in hospital rooms. The profiles of noise levels for their time and frequency domain as being reflected, absorbed, or refracted within typical patient's room are determined. The difference in applications of various architectural solutions to mitigate noise in hospitals within patient environment is estimated. The results contribute toward architectural room acoustic design solution(s) to reduce acoustic disruption of sleep or rest with adequate information and reproducible data to accelerate design decision-making process while providing practical solutions such as geometric modification to room architectural elements as compared to theoretical offerings by the research community.

9:45

2aAAa6. Localization of plumbing noise in a multistory building using an acoustic camera. Marek Kovacik (Scantek, Inc., 6430 Dobbin Rd., Ste. C, Columbia, MD 21045, m.kovacik@scantekinc.com) and Jørgen Grythe (Norsonic, Lierskogen, Norway)

Noise generated by plumbing pipes inside a multi-family dwelling can be a source of great annoyance on its residents. Corrective action of such noise begins with identification of the noise source followed by appropriate noise reduction measures. Source identification becomes more challenging when documentation on the plumbing design is missing or when the noise level is so low that it cannot be localized using practical methods. This paper describes a similar problem where low level noise generated by sanitary pipes is heard inside an apartment building. As a localization tool, an acoustic camera utilizing beam-forming technology is used to identify the noise location inside walls between two adjacent apartment units. Findings are visually presented and discussed.

10:05–10:20 Break

Contributed Papers

10:20

2aAAa7. Gunshot recordings from a criminal incident: Who shot first? Robert C. Maher (Elec. & Comput. Eng., Montana State Univ., 610 Cobleigh Hall, PO Box 173780, Bozeman, MT 59717-3780, rob.maher@montana.edu)

Audio forensic examination for law enforcement and criminal justice investigations increasingly involves audiovisual recordings from dashboard camera systems, bystander smart phones, body cameras worn by police officers, and even by cameras built into TASER^{TM} devices. If the camera is pointing in an appropriate direction the details of the incident may be found in the recorded video. However, if the camera's field of view is limited, it may still be possible to evaluate the circumstances of interest by examining the sounds captured by the recording device's microphone. This paper presents audio examples in which the forensic examiner must attempt to address questions such as: How many gunshots took place? What types of firearms were involved? Who shot first? Audio examples are presented to demonstrate the solutions—and mysteries—found in several real world cases.

10:35

2aAAa8. Noise identification and ongoing research and monitoring for Red Rocks Amphitheatre. Ted Pyper and Matt Whitney (K2 | Consultants in Audio, Video and Acoust., 4900 Pearl East Cir., Ste. 201E, Boulder, CO 80301, ted@k2audio.com)

Beginning in 2013, a noise survey was initiated by the famed Red Rocks Amphitheatre, outside Denver, CO, to respond to noise concerns from nearby communities. Complaints focused on excessive bass frequencies originating from the outdoor venue. Some neighbors even complained of physical vibrations in their residences from amplified music over a mile away. The survey was executed to document the level differences between the sources in the venue and the measured levels in the neighborhoods. Vibration levels were also measured, in addition to airborne noise levels, to verify and diagnose the specific spectrum associated with the complaints. Remedial treatments were studied, and policies have been put in place to monitor and control the overall levels within the venue. Ongoing monitoring and research have evolved over the last several years to better address the concerns of neighbors, while preserving the concert experience within the venue.

10:50

2aAAa9. Shot in the dark: Physical and psychophysical limitations on shooter identification from earwitness testimony. Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

While human listeners have a remarkable ability to determine the cause of a sound and the location of the sound source using only their sense of hearing, this ability is subject to errors and lack of certainty due to factors that are grounded in well-understood phenomena in physics and psychoacoustics. In this talk, the author will describe his experience serving as an expert witness for the defense in a criminal trial concerning an incident that took place in a dark outdoor location in which a shot was allegedly fired by one of two suspects. The defendant was alleged to be the shooter based upon earwitness testimony regarding the direction from which the shot was heard to have originated. Three key questions brought out in the forensic analysis performed for the trial will be addressed from the perspective of signal-detection theory considering both physical factors (e.g., outdoor blast and shockwave propagation) and psychophysical factors (e.g., auditory detection and classification and spatial-hearing acuity): (1) Was the gunshot audible? (2) Was the sound that was heard identifiable as a gunshot? (3) With what accuracy and precision could the location of the gunshot be determined?

Session 2aAAb

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

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

Michael Ermann, Cochair
Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205

David S. Woolworth, Cochair
Oxford Acoustics, 356 CR 102, Oxford, MS 38655

The Technical Committee on Architectural Acoustics of the Acoustical Society of America with support from the Robert Newman Student Award Fund and the National Council of Acoustical Consultants are sponsoring the 2016 Student Design Competition that will be professionally judged at this meeting.

A center for the arts has decided to open a multi-purpose facility. It will include an auditorium with a balcony and platform (stage) to be used as a meeting space and as a music performance chamber. The facility will also include a green room to be used as a meeting room and for music rehearsal. Music performed and rehearsed in this facility will be chamber ensembles and soloists. Instrumentation will be exclusively classical, western orchestral.

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

TUESDAY MORNING, 24 MAY 2016 SALON I, 8:00 A.M. TO 10:10 A.M.

Session 2aAO

Acoustical Oceanography and Signal Processing in Acoustics: Acoustic Consistency of Ocean Models

Timothy Duda, Cochair
Woods Hole Oceanographic Institution, WHOI AOPE Dept. MS 11, Woods Hole, MA 02543

Bruce Cornuelle, Cochair
UCSD-Scripps Institution of Oceanography, 9500 Gilman Drive, Dept. 0230, La Jolla, CA 92093-0230

Chair's Introduction—8:00

Invited Papers

8:05

2aAO1. Multiscale ocean prediction and uncertainty estimation for acoustic studies. Pierre F. Lermusiaux (MIT, 77 Mass Ave., Cambridge, MA 02139, pierrel@mit.edu)

We discuss the prediction and estimation of multiscale ocean fields and their probability density distribution for acoustic studies. In high-fidelity multi-resolution simulations, the probability density function of the full ocean state is predicted and estimated, combining the governing equations with observations. Dynamically balanced stochastic forcing are included, so as to represent effects of sub-grid-scales not resolved by the deterministic model equations. The results are stochastic partial differential equations that allow to capture both deterministic effects (advection, Coriolis, etc.) and statistical effects (smaller-scale turbulence, internal wave variability, etc.) on the environment. With this modeling and data assimilation, accurate estimates of the probability density functions (pdf) of oceanographic variability are possible. They become inputs to end-to-end oceanographic-seabed-acoustic-sonar dynamical systems. This end-to-end uncertainty quantification approach is described, evaluated, and illustrated in several simulations and ocean regions.

8:30

2aAO2. Use of ocean model forecasts during acoustic experiments and naval missions. Kevin D. Heaney (OASIS, Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

Ocean acoustic propagation is critically dependent upon the spatial (and to a lesser extent the temporal) scales of the ocean. With the advent of high-fidelity data-assimilative ocean models, forecasts are available that can provide an improved level of confidence for acoustic propagation modeling in support of sea-tests and naval exercises. In this paper, a set of experiments will be presented where ocean model forecasts were used within an acoustic model to help inform the experiment planning. Tests include shallow water tests (Key West), continental shelf-break environments (Shallow Water 2006, Quantifying and Predicting Uncertainty) and deep water environments (Philippine sea 2009/2010). Models used include the Navy Coastal Ocean Model (NCOM), the MSEAS model (MIT, Pierre Lermusiaux), the ECCO2 model state estimation and the Scripps Institution of Oceanography version of the ROMS (Bruce Cornuelle) model run. *In-situ* assimilation of acoustic signals was performed for geo-acoustic information (in shallow water) and not for ocean model forecast updates. Our conclusion for over 10 years of use of models in real-time experiments is that ocean model forecasting does provide useful information for mission planning and experiment design, particularly when combined with on-site measurements.

Contributed Papers

8:55

2aAO3. Passive ocean acoustic tomography using ships as sources of opportunity recorded on an irregularly spaced free-floating array: A feasibility study. Jacquelyn S. Kubicko, Christopher M. Verlinden (U.S. Coast Guard Acad., New London, CT 06320, jkubicko@gmail.com), Brendan V. Nichols, and Karim G. Sabra (Mech. Eng. Dept., Georgia Inst. of Technol., Atlanta, GA)

This presentation investigates the practical feasibility of using an adaptive volumetric array, such as a series of free-floating buoys with suspended hydrophones, which record ships as acoustic sources of opportunity in coastal waters for performing acoustic thermometry or other environmental inversions in near-shore environments in a totally passive manner. Ships are tracked using the Automatic Identification System (AIS). Numerical simulations using a standard normal mode propagation model were first used to test limitations of the proposed approach with respect to frequency band, drifting receiver configuration, signal to noise ratio, precision, and accuracy of the inversion results, along with sensitivity to environmental and position mismatch. Performance predictions using this model are compared with experimental results using at-sea data collected off the coast of New London, CT, in Long Island Sound during August of 2015.

9:10

2aAO4. A reformulation of the ΛΦ diagram for the prediction of ocean acoustic wave propagation regimes. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu)

The ΛΦ diagram was a tool introduced in the late 1970s to predict ocean acoustic fluctuation regimes termed unsaturated, partially saturated, and fully saturated, where internal wave sound-speed fluctuations play a dominant role. The ΛΦ parameters reflect, respectively, the strength of diffraction and the root mean square phase fluctuation along a ray path. New oceanographic knowledge of the small scale part of the internal wave spectrum, and high angle Fresnel zone formulations now allow a more stable and accurate calculation of these parameters. An empirical relation between the variance of log-intensity and ΛΦ provides a more accurate border between the unsaturated regime and stronger fluctuations. The new diagram is consistent with six short-range, deep-water experiments in the Pacific, Atlantic, and Arctic oceans with frequencies ranging from 75 to 16000 Hz. The utility of the ΛΦ diagram is that it provides one of the few means to inter-compare experiments at different geographic locations, different frequencies, and ranges.

9:25

2aAO5. Ray travel times in the northern Philippine Sea circulation from state estimates. Bruce Cornuelle, Ganesh Gopalakrishnan, Matthew Mazloff, Peter F. Worcester, and Matthew A. Dzieciuch (UCSD-Scripps Inst. of Oceanogr., 9500 Gilman Dr., Dept. 0230, La Jolla, CA 92093-0230, bcornuelle@ucsd.edu)

The North Pacific Acoustic Laboratory (NPAL) Philippine Sea experiment deployed a variety of instruments in the northern Philippine Sea during April 2010 through April 2011, including six acoustic transceivers for reciprocal measurements of travel times between instruments. Five transceivers were moored in a pentagon approximately 660 km in diameter with the sixth transceiver moored in the center. Observed travel-time time series were compared with travel times computed from ocean state estimates made using an eddy-active regional implementation of the MITgcm that were constrained by satellite sea surface height and sea surface temperature observations and by temperature and salinity profiles from Argo, CTDs, and XBTs but not by the acoustic data. The similarities cross-validate the state estimates, while the differences provide a simple estimate of the novel information present in the travel times. Smoothness in the modeled sound speed field had a large effect on the ability to find eigenrays. The ocean state estimates were then re-computed to fit the acoustic travel times. The state estimate was able to match the travel times within their error bars and did not significantly increase the misfits with the other observations.

9:40

2aAO6. The integrated ocean dynamics and acoustics project. Timothy F. Duda, James F. Lynch, Ying-Tsong Lin, Weifeng G. Zhang, Karl R. Helfrich (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), Harry L. Swinney (Univ. of Texas at Austin, Austin, TX), John Wilkin (Rutgers Univ., NB, NJ), Pierre F. Lermusiaux, Nicholas C. Makris, Dick Y. Yue (Massachusetts Inst. of Technol., Cambridge, MA), Mohsen Badiey (Univ. of Delaware, Newark, DE), William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY), Jon M. Collis (Colorado School of Mines, Golden, CO), John A. Colosi (Naval Postgrad. School, Monterey, CA), Steven M. Jachec (U. S. Naval Acad., Annapolis, MD), Arthur E. Newhall (Woods Hole Oceanographic Inst., Woods Hole, MA), Lin Wan (Univ. of Delaware, Newark, DE), Yuming Liu (Massachusetts Inst. of Technol., Cambridge, MA), Matthew S. Paoletti (Univ. of Texas at Austin, Austin, TX), Zheng Gong, Patrick J. Haley (Massachusetts Inst. of Technol., Cambridge, MA), Likun Zhang (Univ. of Texas at Austin, Austin, TX), Kaustubha Raghukumar (Naval Postgrad. School, Monterey, CA), and Michael R. Allshouse (Univ. of Texas at Austin, Austin, TX)

The goal of timely and accurate acoustics modeling in the ocean depends on accurate environmental input information. Acoustic propagation modeling has improved to the point of possibly being ahead of ocean dynamical modeling from the standpoint that some significant ocean features having strong acoustic effects are not faithfully reproduced in many models, particularly data-driven ocean models. This in part stems from the fact that ocean models have developed with other goals in mind, but computational limitations also play a role. The Integrated Ocean Dynamics and Acoustics (IODA) MURI project has as its goals improving ocean models, and also making continued improvements to acoustic models, for the purpose of advancing ocean acoustic modeling and prediction capabilities. Two major focuses are improved internal tide forecasting and improved nonlinear internal wave forecasting, which require pushing the state of the art in data-constrained mesoscale feature modeling as well as developing specialized high-resolution tools. Results are reported on efforts to evaluate internal-tide accuracy in data-constrained models, to insert typically unresolved nonlinear internal waves with nonhydrostatic pressure dynamics into these models, and to make acoustical condition forecasts within three-dimensional operational volumes filled with internal waves.

9:55

2aAO7. Estimating sea-water volume attenuation coefficients from mid-frequency, deep-water experiments in the Atlantic and Pacific. Jeffery D. Tippmann, Jit Sarkar, Chris Verlinden, William S. Hodgkiss, and William A. Kuperman (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, jtippman@ucsd.edu)

A collection of mid-frequency deep-water experiments were performed off the U.S. Pacific coast and in the northeastern Atlantic using a short vertical array cut for 7.5 kHz. Previously presented analysis of a Pacific experiments have shown agreement between experimental estimates of attenuation coefficients to the decades old attenuation models. We present the comparison of experimental attenuation coefficient estimates from additional Pacific and Atlantic data and discuss the methodology for inverting for depth-dependent ocean properties using acoustic amplitude measurements. The data and methodology are also used to explore the feasibility of ocean acoustic attenuation tomography for determining the individual chemical attenuation coefficient terms strictly from *in situ* measurements.

TUESDAY MORNING, 24 MAY 2016 SNOWBIRD/BRIGHTON, 8:00 A.M. TO 9:45 A.M.

Session 2aBAa

Biomedical Acoustics: Acoustic Radiation Force and Elastography

Armen Sarvazyan, Chair
Artann Laboratories, 1753 Linvale-Harbourton Rd., Lambertville, NJ 08530

Contributed Papers

8:00

2aBAa1. Toward validation of shear wave elastography using vibration rheometry in soft gels. Sanjay S. Yengul (Mech. Eng., Boston Univ., 216 Saint Paul St. #201, Brookline, MA 02446, syengul@bwh.harvard.edu), Bruno Madore (Radiology, Brigham and Women's Hospital, Harvard Univ., Boston, MA), and Paul E. Barbone (Mech. Eng., Boston Univ., Brookline, MA)

Shear wave propagation is at the heart of several elastography methods designed to quantify the elastic properties of living tissues. These differ in excitation mechanism, imaging modality, volume of tissue being sampled, and the diagnostic output [Parker *et al.* PMB, 2011]. Some of these techniques are already available to clinicians as diagnostic tools for the identification of pathology, particularly in the diagnosis of liver fibrosis. However, these techniques work in different parameter regimes, making it difficult to quantitatively compare different elastography modalities. As a result, the critical cutoffs for fibrosis staging are system dependent [Ferraioli *et al.* WFUMB, 2015]. We present progress toward a framework to allow quantitative comparison of different viscoelastic measurement techniques, including shear wave based elastography (SWE) methods (e.g., acoustic radiation force based SWE and magnetic resonance elastography), and a novel vibration rheometry of gel phantoms. In so doing, we seek to account for frequency and temperature dependence of material properties, as well as control for other confounding factors such as water content, material aging, boundary conditions, awkward sample geometries, and batch-to-batch gel variations.

8:15

2aBAa2. Potential biomedical applications of non-dissipative acoustic radiation force. Armen Sarvazyan (Artann Labs., 1753 Linvale-Harbourton Rd., Lambertville, NJ 08530, armen@artannlabs.com) and Sergey Tsyuryupa (Artann Labs., Trenton, NJ)

Acoustic radiation force (ARF) has been used for many medical applications, the most notable of which is measuring the stiffness of soft tissues. The ARF used in these applications results from the dissipation of acoustic energy due to scattering and absorption of sound and accordingly may be called dissipative ARF. Another source of ARF, which is not related to dissipation of acoustic energy in medium, is variation in acoustic energy density due to gradients of compressional wave speeds in the medium. The effects of this non-dissipative ARF (nARF) are most pronounced in sonication of medium by short ultrasonic pulses with duration on the order of microseconds. Experiments with tissue-mimicking gelatin blocks and various excised animal tissues demonstrated the possibility of measuring sub-nanometer range displacements induced by nARF acting on interfaces in materials with different compressional wave speeds. A continuous wave Doppler device was used to measure the interface particles velocity. Demonstrated possibility of using Doppler signal processing to extract the nanometer-order motion induced by the nARF could open new areas for ultrasound imaging and tissue characterization. The creation of multi-parametric images of soft tissue using nARF for obtaining maps of the compressional wave speed and shear modulus is considered.

8:30

2aBAa3. Relative contributions of various modes of acoustic radiation force to creating mechanical stress in soft tissues. Armen Sarvazyan (Artann Labs., Lambertville, NJ) and Lev Ostrovsky (Earth System Res. Lab., NOAA, 325 Broadway, Boulder, CO 80305, lev.a.ostrovsky@noaa.gov)

There are several physical effects responsible for the generation of acoustic radiation force (ARF) in soft tissue. The first effect, which is widely used in elastographic applications of ARF, is the transfer of momentum from ultrasonic wave in an attenuating medium due to absorption and scattering. Two other effects include reflection of sound wave from the boundaries between tissues differing in the acoustic impedance and second, spatial variations of the energy density of the propagating wave as a result of variations in the sound wave velocity in a medium. The latter mode of ARF can be directed both outward and toward the source of the propagating wave. In this study, relative contributions of various modes of acoustic radiation force are investigated both theoretically and experimentally. Using Green function for the acoustic potential near the boundaries between tissue structures, the expression for the motion of tissue interfaces under the action of short ultrasonic pulses is derived. Theoretical estimates of dynamics of the soft tissue response to various modes ARF induced by the microsecond range ultrasonic pulses are in agreement with experimental data. Findings of this study may serve as the basis of new approaches in soft tissue elastography.

8:45

2aBAa4. Harmonic modulation of acoustic standing wave fields for tissue engineering. Timothy E. Doyle, Blaine Johnson, Brian D. Patchett (Phys., Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, Timothy.Doyle@uvu.edu), Natalie C. Sullivan (Chemistry, Utah Valley Univ., Orem, UT), Dolly A. Sanjinez, Ashley Behan (Biology, Utah Valley Univ., Orem, UT), Nicole Cowan (BioTechnol., Utah Valley Univ., Orem, UT), and Audrey P. Butler (Univ. of Utah, Salt Lake City, UT)

Acoustic standing wave fields can be used to produce virtual, scaffoldless templates for tissue patterning and engineering. To date, only simple tissue patterns have been attempted and achieved such as layered microstructures. This is due in large part to the use of single-frequency standing waves transmitted from a single source, producing planar nodes and antinodes in the cell/growth medium suspension. The purpose of this study was to explore with computational models the possibility of generating complex tissue microstructures using multiple sources with multiple frequency capabilities. The models simulated multifrequency standing waves (i.e., modulated by higher harmonics), multiple acoustic sources at various angles to each other, and corresponding acoustic cavities. The simulations demonstrated that interference patterns of standing waves could be produced with a three-dimensional complexity similar to that of natural biological structures. Such patterns included alveolar structures (thin, membrane-like walls encapsulating open cavities), lobular structures (cell clusters interlaced with duct- and vascular-like channels), and radial structures (iris-like patterns). Experiments for rendering these models to actual tissue structures have also begun, including the development of multifrequency transducers, using multiple transducers to generate a square lattice of cell walls and channels, and methods to reproduce natural tissue structures with high fidelity.

9:00

2aBAa5. Bone demineralization assessment using acoustic radiation force. Max Denis (Mayo Clinic College of Medicine, 1 University Ave., Lowell, MA 01854, max_f_denis@hotmail.com), Leighton Wan, Mathew Cheong, Mostafa Fatemi, and Azra Alizad (Mayo Clinic College of Medicine, Rochester, MN)

In this work, an ultrasound-guided remote measurement technique is proposed for bone demineralization assessment. Utilizing an acoustic radiation force (ARF) beam as our excitation source and a receiving hydrophone, the mechanical properties of a bone can be noninvasively assessed. Focusing the ARF beam on the bone surface acts as point force generating vibrational waves. Coupling the bone surface and hydrophone, the ensuing radiating acoustic pressure from these vibrational waves are captured for analysis. Of particular interest, are the features best related to the bone's mechanical properties. Conducting *ex-vivo* experiments demonstrated that the velocity feature best delineates intact and demineralized bones. The typical velocity of an intact bone (3000 m/s) is higher in comparison to a 72 h demineralized bone (1600 m/s). According to the receiver operating characteristic (ROC) curve, the optimal velocity cut-off value of 3096 m/s yields 80% sensitivity and 82.61% specificity between the intact and demineralized bones. Other features, such as the spectra of the demineralized bones' acoustic response, exhibited higher attenuation for frequencies below 200 kHz in comparison to the intact bones. A time-frequency analysis demonstrated a frequency shift with demineralization. These results demonstrate the potential application of our proposed technique for monitoring bone demineralization.

9:15

2aBAa6. A generalized reconstruction framework for transient elastography. Mahdi Bayat (Physiol. and Biomedical Eng., Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, bayat.mahdi@mayo.edu), Susanta Ghosh (School of Eng., Duke Univ., Durham, NC), Azra Alizad (Physiol. and biomedical Eng., Mayo Clinic, Rochester, MN), Wilkins Aquino (Dept. of Civil and Environ. Eng., Duke Univ., Durham, NC), and Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN)

Ultrasound transient elastography has emerged as a promising imaging modality for noninvasive evaluation of tissue mechanical properties. Most of the established techniques, however, rely on a pointwise measurement of the induced shear wave speed based on a localized planar wave propagation assumption. This assumption is not always true as most tissues exhibit heterogamous characteristics. Complex interaction of different geometries, boundary conditions, and tissue composition limit the ability of these techniques such that additional spatial-temporal filtering and signal truncations are required. These signal conditionings might lead to improvement in elasticity estimation for some cases (e.g. certain geometries, material properties) but the fact that they do not stem from a comprehensive framework can lead to unrealistic mechanical properties in more complex scenarios. We present a generalized framework based on inverse solution of the elasto-dynamic equations which can simultaneously solve for material properties and geometries without requiring any prior knowledge of the wave types. The method is based on a modified error in constitutive equation (MECE) without requiring additional filtering. We present the results of standard phantom studies based on acoustic radiation force excitation and fast ultrasound tracking where MECE technique successfully estimated the elastic moduli and geometry. [Work supported by NIH grant CA174723.]

9:30

2aBAa7. Differentiation of breast lesions based on viscoelasticity response at sub-Hertz frequencies. Mahdi Bayat, Alireza Nabavizadeh, Viksit Kumar, Adriana Gregory, Azra Alizad, and Mostafa Fatemi (Biomedical and Physiol., Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, bayat.mahdi@mayo.edu)

There is an urgent need for a noninvasive but more accurate method to assess breast masses before referring the patient to biopsy. In this study, we use the slow viscoelastic creep response to evaluate breast masses at low ($<$1 Hz) frequencies. This method uses an automated ramp-and-hold compression device and a high-frame-rate ultrasound imaging to track tissue strain. The study cohort included 30 pre-biopsy patients with suspicious masses. Sequential high-frame-rate images were recorded during the ramp-and-hold process. The data were used to calculate the creep response at each point. Then, using a classic linear model, the retardation-time map was created within the field of view. The resulting retardation-time maps exhibited clear distinction of the mass margins. The results showed that benign breast lesions appear with an increase in creep retardation-time compared to surrounding breast glandular tissue, and the opposite trend was true for the malignant lesions. Statistical analysis of the viscoelasticity features revealed that a contrast feature based on the retardation-time is an accurate classifier of malignant and benign masses ($P<0.0003$). It is concluded that the retardation time is a promising biomarker for differentiation of breast masses. [Work supported by NIH Grant CA168575.]

Session 2aBAb

Biomedical Acoustics: Therapeutic Ultrasound and Microbubbles

Parag V. Chitnis, Chair
Department of Bioengineering, George Mason University, 4400 University Drive, 1G5, Fairfax, VA 22032

Contributed Papers

10:00

2aBAb1. Jet formation of contrast microbubbles in the vicinity of a vessel wall. Nima Mobadersany and Kausik Sarkar (George Washington Univ., 800 22nd St. NW, SEH 3000, Washington, DC 20052, sany@gwmail.gwu.edu)

Behaviors of microbubble contrast agent near a vessel wall under ultrasound excitation are investigated using a boundary integral method. Ultrasound in the presence of microbubbles facilitates drug delivery by streaming and jetting phenomena. The microbubbles are encapsulated by a layer of proteins or lipids to stabilize them against dissolution. The encapsulating shell of the contrast microbubble is viscoelastic modeled here by three different interfacial rheological models. While at low excitations microbubbles undergo regular oscillations, at high amplitudes excitation, they form jets toward the wall. The dynamics and the resulting shear stress on the wall are studied varying the shell rheology and other relevant parameters of the problem.

10:15

2aBAb2. An Eulerian-Lagrangian study of cloud dynamics near a wall. Jingsen Ma, Georges L. Chahine, and Chao-Tsung Hsiao (Dynaflow, Inc., 10621-J Iron Bridge Rd., Jessup, MD 20794, jingsen@dynaflow-inc.com)

The dynamics of bubble clouds is studied using an Eulerian-Lagrangian model treating the two-phase medium as a continuum and modeling the microbubbles as discrete sources and dipoles tracked in a Lagrangian fashion. These two are coupled through the local void fractions associated with the instantaneous bubble volumes and locations. Resonance of the bubble cloud, resulting in the highest pressure at the nearby rigid wall, is deduced from simulations with the same initial bubble distribution while varying the excitation frequency and amplitude. This resonance frequency deviates significantly from the classical linearized solution as the relative amplitude of driving pressure increases. It gradually drops as the excitation amplitude increases until reaching a limit value. In the high amplitude driving pressure regime, the peak pressure generated at the wall has a maximum for an optimum value of the initial bubble radius for the same cloud radius. This occurs when the ratio of maximum over initial bubble size is maximum. Too weak or too strong bubble interactions in the cloud inhibit strong collective effects. For the same initial void fraction and for different initial bubble sizes (changing the bubble number), an optimum bubble size also exists, resulting in the highest collective pressure.

10:30

2aBAb3. Optical property changes in *ex vivo* tissues exposed to high-intensity focused ultrasound. Jason L. Raymond, Eleanor Edwards, Robin O. Cleveland, and Ronald A. Roy (Dept. of Eng. Sci., Univ. of Oxford, 17 Parks Rd., Oxford OX1 3PJ, United Kingdom, jason.raymond@eng.ox.ac.uk)

Real-time acousto-optic (AO) sensing has been shown to non-invasively detect changes in *ex vivo* tissue optical properties during high-intensity focused ultrasound (HIFU) exposures. Baseline changes in optical properties have been previously measured as a function of thermal-dose for tissues exposed to a temperature-controlled water bath (doi:10.1088/0031-9155/59/13/3249). In this work, the wavelength-dependent optical scattering and absorption of *ex vivo* tissues exposed to HIFU were measured using an integrating sphere spectrophotometric technique employed previously. HIFU-induced spatiotemporal temperature elevations were measured using an infrared camera and used to calculate the thermal dose delivered to a localized region of tissue. We consider the impact of thermal dose, temperature elevation, and heating rate on the formation of HIFU lesions and the resulting changes in tissue optical properties. Recently, reported results (doi:10.1088/0031-9155/59/13/3249) refute the hypothesis that optical property changes in tissue are based solely on accumulated thermal dose—this claim will be explored further through the optical characterization of lesions formed in clinically relevant tissues. Results will show how wavelength-dependent optical property changes in tissues can be used to improve the AO sensing of lesion formation during HIFU therapy as an alternative to thermometry. [Work supported by the ASA F.V. Hunt Fellowship and the University of Oxford.]

10:45

2aBAb4. Tracking kidney stones during shock wave lithotripsy. Kya Shoar (Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, kya.shoar@magd.ox.ac.uk), Ben Turney (Nuffield Dept. of Clinical Medicine, Univ. of Oxford, Oxford, United Kingdom), and Robin Cleveland (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Shock wave lithotripsy is a non-invasive procedure by which kidney stones are fragmented by shock waves. Real-time monitoring during shock wave lithotripsy remains a challenge more than three decades after it was introduced into clinical practice. Currently, many shock waves are delivered to the body that do not impact the stone, but do result in tissue trauma. This work presents a monitoring system to locate kidney stones, with the goal of gating shock waves not aligned with the stone, and hence, reduce renal trauma during lithotripsy. A circular array, housing twenty-two 0.5 MHz transducers that can be mounted on a clinical lithotripter, was deployed in a water tank. An algorithm, consisting of threshold detection, including automatic rejection of weak signals, and triangulation, was developed to determine the location of stones. The accuracy of the system was tested using: a spherical steel ball and two stone models made from gypsum cement. The results show that within ± 15 mm of the focus of the lithotripter, the accuracy was better than 5 mm in the lateral directions and 2 mm in the axial direction. Using off-the-shelf hardware, the algorithm can calculate stone position every 1.1 s, allowing for real-time tracking during lithotripsy. [Work supported in part by NIH through P01-DK43881.]

11:00

2aBAb5. Effects of acoustic parameters on nanodroplet vaporization. Krishna N. Kumar, Mitra Aliabouzar, and Kausik Sarkar (George Washington Univ., 800 22nd St. NW, SEH 3000, Washington, DC 20052, krishnasst@gmail.com)

Phase shift nanodroplets offer a number of advantages over ordinary microbubbles due to their enhanced stability and smaller size distribution. These nanodroplets undergo a phase transition from liquid to highly echogenic gaseous state when activated by sufficient acoustic energy through a process termed acoustic droplet vaporization (ADV). In this study, we synthesized lipid-coated perfluoropentane (PFP) filled nanodroplets via sonication

method. We investigated the ADV threshold of these nanodroplets as a function of acoustic parameters such as excitation pressure, frequency, pulse length, and pulse repetition period (PRP). Our results indicate that ADV threshold varies significantly with PRP; while at PRP of 10 ms, the ADV threshold was found to be 3.6 MPa (pk-pk), for PRP of 1 ms, 100 μs, and 500 μs, ADV was not observed even at 10 MPa. At ADV, fundamental and odd harmonics were found to be significantly higher than the background noise. The acoustic response of ordinary perfluorobutane (PFB) filled microbubbles with the same lipid composition is compared to that of PFP nanodroplets when both were excited with the same excitation pressures (450 kPa to 10 MPa). Above ADV threshold both showed a similar response.

11:15

2aBAb6. Effects of ultrasound in presence of microbubbles on cartilage tissue regeneration in three-dimensional printed scaffolds. Mitra Aliabouzar, Lijie G. Zhang, and Kausik Sarkar (George Washington Univ., 800 22nd St. NW, SEH 3000, Washington, DC 20052, mitra.aalee@gmail.com)

Gas-filled microbubbles encapsulated with lipids and other surfactants are highly responsive to ultrasound, which has led to their effective role as ultrasound contrast agents (UCA). In this study, for the first time, we used lipid-coated microbubbles (MB) prepared in-house in order to better harness the beneficial effects of ultrasound stimulation on proliferation and chondrogenic differentiation of human mesenchymal stem cells (MSCs) within a novel 3D printed poly (ethylene glycol) diacrylate (PEG-DA) hydrogel scaffolds. A significant increase in cell number (p <0.001) was observed with low intensity pulsed ultrasound (LIPUS) treatment in the presence of 0.5 % (v/v) MB after 1, 3, and 5 days of culture. MSC proliferation enhanced up to 40% after 5 days of culture in the presence of MB and LIPUS while this value was only 18% when excited with LIPUS alone. We investigated the effects of acoustic parameters such as excitation intensity, frequency, and pulse repetition period on MSC proliferation rate. Our 3-week chondrogenic differentiation results demonstrated that combining LIPUS with MB significantly enhanced both Glycosaminoglycan (GAG) and type II collagen production. Therefore, integrating LIPUS and MB appears to be a promising strategy for enhanced MSC growth and chondrogenic differentiation.

11:30

2aBAb7. Microbubble response to dual frequency excitation for broadband contrast imaging. Christina Keravnou (Dept. of Bioeng., Univ. of Washington, Benjamin Hall University of Washington, 616 NE Northlake PL., Seattle, WA 98105, ckerav@uw.edu), Chrysovalantis Papantonis (Dept. of Mech. and Manufacturing Eng., Univ. of Cyprus, Nicosia, Cyprus), and Michalakis Averkiou (Dept. of Bioeng., Univ. of Washington, Seattle, WA)

Contrast imaging relies on pulsing schemes like pulse inversion (PI), power-modulated pulse inversion (PMPI), and power modulation (PM) to isolate certain harmonic components. These multi-pulse schemes normally employ single frequency pulses (SFP). In this work, we propose dual frequency pulses (DFP) coupled with pulsing schemes (PI, PMPI, and PM) for microbubble imaging. The proposed DFP contain an amount b_1 of a fundamental frequency f and an amount b_2 of its phase-shifted (φ) second harmonic 2f. Microbubbles were excited with a variety of DFP combinations (b_1, b_2, φ) and their nonlinear response was compared to that from conventional SFP. All pulsing schemes benefit in terms of overall signal from the use of DFP by ~4–6 dB. PI benefits the most when coupled with DFP as it extracts an additional amount of nonlinear fundamental signal (~25 dB) something not feasible with SFP. It was observed that microbubble nonlinear response changes when the relative amount of f and 2f in DFP change, allowing the design of specific pulses for different microbubbles. In conclusion, since the use of DFP with pulsing schemes extracts harmonics more efficiently than SFP it has the potential to improve contrast imaging in terms of bubble sensitivity and image resolution.

11:45

2aBAb8. Lytic efficacy of tissue plasminogen activator and ultrasound in porcine clots doped with barium sulfate *in vitro*. Shenwen Huang, Himanshu Shekhar, and Christy Holland (Univ. of Cincinnati, 231 Albert Sabin Way, Mail Location: 0586, Cincinnati, OH 45229, huangsw@mail.uc.edu)

Swine and porcine tissue are employed routinely for preclinical models of ischemic stroke. In this study, we examined the lytic susceptibility of porcine whole-blood clots, prepared with and without barium sulfate, a radiopaque x-ray contrast agent. The degree of lytic efficacy in the two types of clots was compared for recombinant tissue plasminogen activator (rt-PA) concentrations ranging from 0 to 100 μg/mL. Subsequently, a *in vitro* flow model and time-lapse thrombolysis microscopy system was used to evaluate lysis in clots with and without barium sulfate in response to rt-PA and 120-kHz intermittent ultrasound exposure. The degree of lysis observed with both types of clots was similar for rt-PA concentrations up to 15.75 μg/mL. However, clots doped with barium sulfate demonstrated significantly lower lysis at higher rt-PA concentrations. Similarly, results obtained using the *in vitro* flow model showed that both types of clots underwent comparable lysis when treated with 15.75 μg/mL rt-PA. Further, using Definity® and 120-kHz ultrasound as an adjuvant to rt-PA did not enhance lysis in either porcine clot model compared to rt-PA alone. These results highlight the considerable differences in the degree of clot lysis between porcine clots and previously reported studies that used human clots.

Session 2aEA

Engineering Acoustics: General Topics in Engineering Acoustics I

Kenneth M. Walsh, Chair
K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Chair's Introduction—8:00

Contributed Papers

8:05

2aEA1. Characterization of porous material and a sound suppression system using a shock tube. Erica K. Good, Robert Taylor, James Luskin, and Zachary Conaway (College of Eng., The Catholic Univ. of America, 655 Michigan Ave. NE, Apt. P502, Washington, DC 20017, 45taylor@cua.edu)

This work describes the design and construction of an experimental apparatus for testing shock waves and a sound suppression system for student learning. The shock tube will be used to characterize both porous media and the suppression system response to a non-linear shockwave. The effects of a plane shock wave will be used to determine acoustic properties, such as transmission, reflection, and absorption coefficients, of rigid frame porous materials. The suppression system is comprised of 20 conical chambers that cause a loss of energy in the shock wave by the expansion and acceleration of the flow. The sound pressure level from the shock is in excess of 140 dB re 20 μPa. Material properties and the performance of the designed suppression system will be presented.

8:20

2aEA2. A method to suppress harmonics in standing-wave thermoacoustic engines. A. H. Ibrahim, M. Emam (American Univ. in Cairo, New Cairo 11835, Egypt, abdelmaged@aucegypt.edu), A. M. Fouad (Mech. Power Dept., Cairo Univ., Giza, Giza, Egypt), and Ehab Abdel-Rahman (American Univ. in Cairo, New Cairo, Egypt)

Harmonic generation in thermoacoustic engines is regarded as a non-linear loss mechanism that extracts acoustic power from the fundamental wave into harmonics. It is shown that the use of specifically designed insert that limits the gas flow area over a very-limited part of the resonator can suppress this non-linear loss mechanism causing a significant increase in the generated acoustic power in the fundamental mode. In this experimental work, a standing-wave thermoacoustic engine is built and operated without inserts and with inserts of different shapes, porosities, and thicknesses. The self-generated dynamic pressure waves are captured under different operating conditions and then are decomposed into a fundamental component and harmonics. Results for different inserts are presented and discussed. All inserts caused lower harmonic content with respect to the no-insert case but inserts of low gas flow area cause severe flow blockage and thus a severe reduction in the produced acoustic powers. Results analyze the relationships between the generated acoustic power in the fundamental mode and the amplitudes of the dynamic pressures of the fundamental, first harmonic, and second harmonics. The blockage effects caused by the insert are discussed.

8:35

2aEA3. Efficient dispersion analysis of guided waves in laminated composites and substrates. Ali Vaziri and Murthy N. Guddati (Civil Eng., North Carolina State Univ., 208 Mann Hall, 2501 Stinson Dr., Campus Box 7908, Raleigh, NC 27695-7908, avaziri@ncsu.edu)

Nondestructive testing of laminated composites and substrates involve matching observed and predicted dispersion characteristics of guided waves. These characteristics are quantified through dispersion curves (phase velocity versus frequency) and need to be computed for a large number of estimated structure and material property combinations. Given its central nature, we propose an efficient approach for computing the dispersion curves, leading to an order of magnitude savings in the computational cost. Our approach is based on conventionally used finite element semi-discretization through the depth, but with one significant modification: by using a specially designed set of complex-valued finite element lengths through the depth, we show that the dispersion curves can be obtained with a handful of elements per layer as opposed to larger number of traditional finite elements, resulting in large reduction in computational effort. In this talk, we present the formulation of the proposed complex-length finite element method and illustrate its efficiency through modeling wave dispersion in laminated composites and substrates. Finally, we introduce an inversion procedure developed around this method and demonstrate its effectiveness in characterizing plate structures using synthetic as well as real nondestructive testing data.

8:50

2aEA4. Reverberation characterization inside an anechoic test chamber at the weapon sonar test facility at Naval Undersea Warfare Center Keyport Division. Grant C. Eastland and William C. Buck (Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowell St., Keyport, WA 98345, grant.eastland@navy.mil)

The Weapon Sonar Test Facility (WSTF) at NUWC Keyport, WA, is a 34500 gallon pressure tank currently used for test and evaluation of torpedo sonar arrays. The tank has many more possible uses and complete characterization of the testing environment needs to be performed. One of the methods of characterization being used is the determination of the reverberation time of the anechoic chamber. Applying Sabine-Franklin-Jaeger theory of reverberant rooms, an experimentally determined reverberation time T_{60} for a "live room" can be used to provide an upper bound for the reverberation time in the chamber. Utilizing the Eyring theory of "dead" rooms will provide a better determination of T_{60} for the anechoic chamber, hence provide a lower bound characterization. The experimental method involves determining the spatially averaged acoustic energy of an initial tone and the corresponding reflection from the chamber wall seen in a recorded signal. These values determined from the root mean square signal voltage determine the wave decay time, τ. The decay time is directly related to the reverberation time through $T_{60} = 6\tau\times\log_e 10$. The reverberation was determined to be within an acceptable tolerance for testing and evaluation.

9:05

2aEA5. A comparison of different methods for calculating complex acoustic intensity. Eric B. Whiting, Kent L. Gee, Scott D. Sommerfeldt, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, ericbenwhiting@gmail.com)

The Phase and Amplitude Gradient Estimation (PAGE) method of calculating acoustic intensity from multiple pressure measurements [Thomas *et al.*, JASA **137**, 3366–3376 (2015)] has been used successfully to calculate active intensity in the laboratory and in-field experiments. The primary result is that the PAGE method increases the bandwidth over which accurate vector intensity estimates can be obtained. Further work has investigated the application of the PAGE method to calculate other energy-based quantities, including the reactive intensity. Simulated pressure and particle velocity fields have been calculated for a single monopole, two out-of-phase monopoles, and, via Rayleigh integration, a baffled circular piston. The analytical intensity fields are then compared to the PAGE and the traditional p-p methods for obtaining both the active and reactive intensity, both in the near and far fields. [Work supported by NSF.]

9:20

2aEA6. Study of virtual instrument technology applied in sound field test. Yongchao Yao, Xiaodong Ju, Junqiang Lu, and Baiyong Men (China Univ. of Petroleum (Beijing), No.18, Fuxue Rd., Changping District, Beijing, Beijing 102249, China, Wardruna2013@163.com)

As a widely used technology in the field of test and measurement, virtual instrument (VI) combines high-performance modular hardware with flexible software to perform a variety of experimental tasks. In order to perform the sound field test in laboratory, we build an specialized platform of VI, whose hardware system mainly consists of multi-channel data acquisition boards, independence analyzer, mechanical positioning device, and industrial personal computer equipped with PXI, GPIB, and Ethernet bus interface. Also, a graphical VI programming language (LabVIEW) is adopted to develop the corresponding software system. With this VI platform, we measure the impedance property of transducers and the sound field of a phased array composed of them. The results show that the transducers are positioned correctly by the mechanical positioning device, the excitation time of every transducer of the phased array is controlled accurately, and finally a set of high-quality waveforms are acquired.

9:35

2aEA7. Loudspeaker system. Gordon Pisani, Nico Bartolomeo, Joseph Mascolo, Gabrielle Wolff, Alexander Laprade, Ethan Gorczynski, and Diego Turo (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC 20064, 17pisani@cua.edu)

This presentation will demonstrate the design, construction, testing, and final performance of a student designed loudspeaker system. The purpose of this project was to create a 2.1 channel, high quality sound system that produces a bandwidth just short of the full range of human hearing, approximately 20–20000 Hz. This system consists of 2 two-way speakers utilizing a woofer and tweeter, with one separate subwoofer. The function of this particular sound system is to reproduce sound faithfully in a 350–400 square foot room. The performance characteristics of this system will be presented through frequency response and linearity testing results. These tests were used to calculate and implement modifications made to the cabinet size and shape while adhering to a budget of $1200. The analysis of every performance will also reveal data for different electrical component values in designing the Linkwitz-Riley crossovers.

9:50

2aEA8. Jet pump oscillating-flow behavior at different Womersley numbers. Abdelrahman Nassif, Ahmed I. Abd El-Rahman, and Ehab Abdel-Rahman (Phys., American Univ. in Cairo, AUC Ave., New Cairo 11835, Egypt, ahmedibrahim@aucegypt.edu)

Few numerical works simulate the *laminar*-flow behavior within the diverging passages of typical thermoacoustic jet pumps. The associated boundary-layer separation is mostly delayed by maintaining a slowly increasing flow passage and thereby the corresponding pressure drop and consumed acoustic energy are noticeably reduced. Of equal importance is the effect of the *transition to turbulence* on the boundary-layer separation and the resulting flow-pattern. To our knowledge, no simulation has been developed that numerically predict and capture the conditionally turbulent flow regime within typical jet pumps. Here, a new finite-volume model is reported that employs the non-linear CFD solver of ANSYS FLUENT. The k-kl-ω transition model is considered and its parameters are particularly adjusted for present application. A sinusoidal pressure oscillation is enforced at one end, while the far-field approximation is considered at the other end to model the non-reflecting boundary condition. An axisymmetric model along with careful meshing are applied. The transient simulation run proceeds till stationary flow behavior is obtained. Both the flow streamlines and vorticity field are plotted for three different acoustic frequencies, Womersley numbers of which correspond to 5, 15, and 30. The numerical results capture the onset of flow transition into turbulence and characterize the flow patterns at different frequencies. The results also illustrate the influence of the developed turbulent flow-patterns on the boundary-layer separation.

10:05–10:20 Break

10:20

2aEA9. Classification and optimization of beam responses synthesized from non-uniformly spaced linear arrays. Chrisna Nguon (Univ. of Massachusetts Lowell, Dracut, MA), Nicholas Misiunas, Jenny Au, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, Nicholas_Misiunas@student.uml.edu)

A probability distribution for the element positions on a linear array is derived so as to match a target beam pattern that approximates a Dolph-Chebychev array. The random beam responses generated from such a configuration exhibit a mean profile in observation angle that is invariant with element size N but the variance decays as 1/N. The beam responses are classified considering the N-1 inter-element distances as features that control the deviation from the mean beam profile and the variance of the class. The inter-element distance constraints are incorporated in a computational model of the firefly algorithm (FA) designed to optimize the location of the array elements that match the target beam pattern. The FA consists of a set of fireflies, representing arrays, which move through the solution space with directed movement based on each fireflies' fitness and random movement to ensure adequate exploration and avoid local minima. The fitness function is a combination of a target beam-width, side-lobe level, and bounds on the position dependent mean inter-element distances. The variance of the resulting beams are compared with the result from the unconstrained random placement of elements as a function of the element size and side-lobe parameters.

10:35

2aEA10. Analysis of the error sources of the two-microphone transfer function method for measuring absorption coefficient in the free field using numerical modeling. Hubert S. Hall, Joseph F. Vignola, John A. Judge, and Diego Turo (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC 20064, 61hall@cua.edu)

Extension of the two-microphone transfer function absorption coefficient measurement technique from impedance tubes to the free field introduces several error sources. The three-dimensional nature of the sound field necessitates consideration of factors that are not relevant in an enclosed tube below the cutoff frequency. The impedance tube technique has been modified to account for non-planar wave propagation due to an acoustic point source. The sound field contamination from sample edge diffraction has generally restricted use of the technique to frequencies such that wavelengths are small relative to sample dimensions. This requires the use of very large test panels for low frequencies. Numerical models of the two-microphone free field technique have been created to quantify these effects. Each effect was isolated to better understand its independent impact on the accuracy of the technique. Finally, a series of experimental tests were conducted to validate the numerical modeling results.

10:50

2aEA11. Optimizing an underwater transmit array of projectors with large variations. Eunghwy Noh, Wonjong Chun, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, ohm@yonsei.ac.kr), and Youngsoo Seo (Agency for Defense Development, Changwon, Gyeongsangnam-do, South Korea)

Uniformity of constituent projector elements is a prerequisite for building a high-performance underwater transmit array. Unless the element-to-element variations in transmitting voltage response (TVR) and directivity fall within specified limits, the resulting acoustic performance of the array will suffer significantly in terms of its sound output and directivity. Here, we consider a case in which projector elements exhibit a large scatter in TVR, say, over 6 dB, and present a design guide to still get the most of the array. The focus of the study is to explore how the relative placement of projector elements affects their mutual interaction and ultimately leads to the overall characteristics of the array. Finite element computations are carried out to derive an optimized design and the design guide. [This work has been supported by the Low Observable Technology Research Center program of Defense Acquisition Program Administration and Agency for Defense Development.]

TUESDAY MORNING, 24 MAY 2016 SALON I, 10:25 A.M. TO 11:55 A.M.

2a TUE. AM

Session 2aED

Education in Acoustics: Education in Acoustics General Topics

Jon P. Johnson, Cochair
Physics, Brigham Young University - Idaho, 525 S Center St., Rexburg, ID 83460-0520

Ryan Nielson, Cochair
Brigham Young Univ. - Idaho Physics, 525 S. Center St., Rexburg, ID 83460-0520

Contributed Papers

10:25

2aED1. Cultivating successful undergraduate research by cognitive apprenticeship: A case study. Teresa J. Ryan (Dept. of Eng., East Carolina Univ., 248 Slay Hall, Greenville, NC 27858-4353, ryante@ecu.edu)

A common challenge in academic research is finding a meaningful and productive way to involve undergraduate students in substantive research activities. The Department of Engineering at East Carolina University offers a general Engineering undergraduate degree, and matriculated its first small group of graduate students into a Master's program in Biomedical Engineering in Fall of 2014. With few master's students and no doctoral program, our undergraduate students are often given the opportunity to play a role in research that is more typically expected of a traditional graduate research assistant. This fact presents both great opportunity and unique challenges. This work presents the framework for a case study in cognitive apprenticeship that will take place over the next four years. The group of students includes all class levels, from freshmen to seniors. In addition, implementation of a hive or collaborative approach for introducing undergraduate students to advanced technical literature is described.

10:40

2aED2. An open-access interactive textbook for teaching room acoustics modeling. Lauri Savioja (Dept. of Comput. Sci., Aalto Univ., PO Box 15500, Aalto FI-00076, Finland, Lauri.Savioja@aalto.fi)

There are several different techniques for modeling room acoustics. Interactive visualizations can be used to ease learning the main principles and algorithms for that. This presentation will demonstrate a new interactive open-access textbook aiming to support learning the basics of room acoustics modeling techniques. The visualizations illustrate how different algorithms work in simple 2-D geometries, and what are the underlying assumptions in each of them. Interactivity here means that the students have control over both various parameters that affect the modeling performance and accuracy, and the geometries. They can study how the modeling results change in response to such changes. The main emphasis here is on geometrical room acoustics, and the techniques based on that: image-source model, ray-tracing, and acoustic radiance transfer, but the longer term aim is to cover all the main room acoustic modeling principles and techniques.

10:55

2aED3. The sound of STEAM (science, technology, engineering, arts, and math): Acoustics as the bridge between arts and STEM (science, technology, engineering, and math). Caleb Goates, Jenny Whiting (Brigham Young Univ., N283 ESC, Provo, UT 84602, calebgoates@gmail.com), Mark Berardi (Michigan State Univ., East Lansing, MI), Kent L. Gee, and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

This paper describes the development and presentation of a Science, Technology, Engineering, Arts, and Math (STEAM) workshop for elementary school teachers designed to provide ideas and tools for using acoustics in the classroom. The abundant hands-on activities and concepts in acoustics naturally link science and music in an intuitive way that can assist teachers in moving forward on the STEAM initiative. Our workshop gave teachers an introduction to acoustics principles and demonstrations that can be used to tie STEAM techniques in with Utah State Education Core standards. These hands-on demonstrations and real-world applications provide an avenue to engage students and support learning outcomes. Feedback indicated that the participants learned from and enjoyed the initial implementation of this workshop, though many elementary school teachers did not immediately see how they could integrate it into their curriculum. While additional efforts might be made to better focus the training workshop for the K-6 level, curriculum developers need to appreciate how acoustics could be used more broadly at the elementary school level if the emphasis changes from STEM to STEAM.

11:10

2aED4. Loudspeaker design and analysis. Joseph Mascolo (Mech. Eng., The Catholic Univ. of America, 38 Canfield Rd., Morristown, N J 07960, 30mascolo@cua.edu), Nico Bartolomeo, Alexander Laprade, Ethan Gorczynski, Gabrielle Wolff, Gordon Pisani, and Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This presentation will demonstrate the planning, construction, testing, and final performance of a student designed loudspeaker system. The purpose of this project was to create a 2.1 channel, high quality sound system that produces a bandwidth just short of the full range of human hearing, which is then cut off and sent to the appropriate speakers. This system consists of 2 two-way speakers utilizing a woofer and tweeter, arranged in unison with one sole subwoofer. The function of this particular sound system is to reproduce sound faithfully in a 350–400 square foot room, yet still be lightweight and maneuverable. The performance characteristics of this system will be presented through frequency response and linearity testing results. These tests were used to help warrant modifications made to the cabinet size and shape while adhering to a strict budget of $1,200. The analysis of every performance will also reveal data for different electrical component values in designing the Linkwitz-Riley crossovers. The aim of this presentation is to highlight the various key factors that are important in loudspeaker design.

11:25

2aED5. High frequency sound speed measurements in liquids and solids using a smartphone. Blake T. Sturtevant and Dipen N. Sinha (Mater. Phys. and Applications, Los Alamos National Lab., MPA-11, MS D429, Los Alamos, NM 87545, bsturtev@lanl.gov)

Much has been written in recent years on the use of smartphones and other personal electronic devices (PED) in the high school and college teaching laboratories. Due to the bandwidth limitations ($<$ 20 kHz) of these devices, experiments proposed to date have focused almost exclusively on measurements of phenomena in air and in the audible range. This talk will focus on a simple way to extend the use of these devices to measure sound speed in liquid and solids which have significantly higher sound speeds than air. High frequency acoustic standing waves in these media can be generated using standard teaching laboratory instrumentation (e.g., signal generators providing white noise, chirp signals, CW signals with manually adjusted frequency, etc.) and two opposing piezoelectric disks. By down-converting the high frequencies with an inexpensive diode mixer and measuring the difference frequency spectrum with a PED, it is possible to determine multiple successive standing wave frequencies in the medium. The measured frequencies, along with the distance between the disks, can then be used to determine sound speed accurately. Simple and inexpensive, the proposed system provides a learning experience traditionally inaccessible without research grade RF measurement instrumentation.

11:40

2aED6. A new acoustics research group and course at Brigham Young University—Idaho. Jon P. Johnson (Phys., Brigham Young Univ. - Idaho, 525 S Ctr. St., Rexburg, ID 83460-0520, johnsonjo@byui.edu), Kent L. Gee (Phys., Brigham Young Univ., Provo, UT), and Ryan Nielson (Phys., Brigham Young Univ. - Idaho, Rexburg, ID)

This paper details the beginning of a collaboration between the Acoustics Research Group at Brigham Young University in Provo, UT, and a new research group at BYU-Idaho in Rexburg, ID. BYU-Idaho is a four-year university without a graduate program. Its physics program awards 10 to 15 BS degrees in physics each year. Historically, only a 100-level acoustics course has been offered by the physics department, but in conjunction with forming this new research group we are hoping to construct an upper-division acoustics course assuming a background in differential equations, computational physics, and Fourier analysis. An overview of the collaboration, which is still in its infancy, is discussed.

TUESDAY MORNING, 24 MAY 2016 — SALON B/C, 8:00 A.M. TO 10:50 A.M.

Session 2aMU

Musical Acoustics: Voice Registration in Amplified and Unamplified Singing

Ingo R. Titze, Chair
National Center for Voice and Speech, 136 South Main Street, Suite 320, Salt Lake City, UT 84101-3306

Chair's Introduction—8:00

Invited Papers

8:05

2aMU1. The affect of audio enhancement on vocal timbre. Matthew Edwards (Voice/Musical Theatre, Shenandoah Univ., 129 Morning Glory Dr., Winchester, VA 22602, medwards09@su.edu)

In acoustic singing, performers learn to finely tune their resonance in order to control the spectral output of their voices. However, in amplified singing styles, microphone frequency response, equalization, and other editing tools can affect our perception of resonance. These effects can have positive and negative impacts on the spectral output of a singer's voice that teachers must be aware of when training Contemporary Commercial Music (CCM) artists. This presentation will include information on how audio equipment can alter the timbre of the voice, affect our perception of registration, and alter the singing power ratio.

8:35

2aMU2. Nonlinear dynamics helps explain how vowel influences register stability. Lynn M. Maxfield (National Ctr. for Voice and Speech, Univ. of Utah, 136 S Main St., Ste #320, Salt Lake City, UT 84101, lynn.maxfield@utah.edu)

Highly trained singers know what a registration shift feels like, where in their ranges it is likely to occur, and how to make small adjustments to make it less abrupt. They also know that the location of register changes differs depending on vowel choice and the style in which they are performing. Changing registers while singing can result in a significant timbral shift and, if approached unexpectedly, an abrupt jump in f_o. If the source and filter interact in a purely linear fashion, f_o should not be influenced by the shape of the vocal tract (vowel). This paper will demonstrate how nonlinear source-filter coupling may explain the strong relationship between vowel and registration. Eight volunteers performed f_o glides while altering the dimensions of their vocal tracts, predictably changing the formant frequencies. It was hypothesized that if the source and filter operated as a purely linear system, f_o stability should not be perturbed by formant/harmonic crossings in a linear system. Acoustic analysis revealed, however, that 85% of f_o instabilities were aligned with a crossing of one of the first four harmonics with the first three formants, indicating that source-filter coupling likely occurred.

9:05

2aMU3. Why fry? An exploration of the lowest vocal register in amplified and unamplified singing. John P. Nix (Music, Univ. of Texas San Antonio, One UTSA Circle, San Antonio, TX 78249, john.nix@utsa.edu)

Vocal fry is the lowest register in human vocal production. It can be defined on a psycho-acoustical basis as vocalization at fundamental frequencies below the perceptual threshold for discrete events, which occurs at approximately 70 Hz. Vocal fry has become more commonplace in conversational speech and amplified singing styles such as popular and country, but it is typically unused in non-amplified accompanied performances of most Western Classical music. The author's presentation will include the results of a survey of listener preferences of performances of popular and country performances with and without vocal fry, and the results of an experiment to examine what acoustical information is being transmitted to listeners during the fry portions of performances.

9:35

2aMU4. Vocal registers and comparative frequency analysis for multiple sung genres. Lisa Popeil (Voiceworks, 14162 Valley Vista Blvd., Sherman Oaks, CA 91423, lisapopeil@mac.com)

Diverse vocal genres utilize different registration and resonance strategies. This presentation will analyze and compare sound samples from numerous sung genres including classical (male versus female), pop, R&B, rock, country and various musical theater belting substyles (heavy belt, nasal belt, ringy belt, brassy belt, and speech-like belt). In addition, using acoustic analysis, these styles will be demonstrated by a single subject to show how registration and resonance choices can be made to clarify, identify, and codify the conventions of that genre.

10:05–10:20 Break

Contributed Papers

10:20

2aMU5. Listener ratings of singer expressivity in musical performance. Mackenzie L. Parrott and John Nix (Music Dept., Univ. of Texas at San Antonio, 1 UTSA Circle, San Antonio, TX 78249, mackenzie.lanae@gmail.com)

Vocal fry has gained a lot of attention in recent years from speech language pathologists, linguists, singers, researchers, and the general public alike. Vocal fry register uses low airflow through shortened vocal folds, causing an aperiodic pattern in vocal fold oscillation. This vocal mannerism is becoming increasingly common in American speech, fueling discussion about the implications and perception of its use. As it has become more prevalent, fry has naturally found its place in many commercial American song styles as well. Many singers are implementing fry as a stylistic device at the onset or offset of a sung tone. The objective of this study is to analyze whether listeners' ratings of a singer's expressivity in musical samples in two contemporary commercial styles (pop and country) are affected by the presence of vocal fry. We hope that this study will shed some insight into the prevalence of this particular vocalism in popular music, and to see if there is a difference in listener ratings according to the singer's gender.

10:35

2aMU6. Toward mapping voice registers in the physiological domain: A computational study. Zhaoyan Zhang (UCLA School of Medicine, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Although there have been many studies on the acoustics and vibration of voice registers, the underlying physiological mechanisms of different registers still remain unclear. This study represents a preliminary effort toward identification of regions in the physiological domain that would lead to different voice registers, using a computational phonation model. The physiological parameters under consideration include vocal fold geometry, stiffness, resting glottal angle, subglottal pressure, and vocal tract shapes. Three registers are considered, including vocal fry, chest, and falsetto. Voice registers are identified based on known or reported acoustic features, including F0, closed quotient, and harmonic structures, based on which the physiological conditions corresponding to each voice register are determined. Preliminary perceptual experiment will be performed to confirm register identification. [Work supported by NIH.]

Session 2aNSa

Noise, Signal Processing in Acoustics, Architectural Acoustics, and ASA Committee on Standards: Noise Measurements with Mobile Apps

Benjamin Faber, Chair
Faber Acoustical, LLC, 277 S 2035 W, Lehi, UT 84043

Chair's Introduction—8:00

Invited Papers

8:05

2aNSa1. Evaluation of smartphone sound measurement applications using external microphones—A follow-up study. Chucri A. Kardous, Peter B. Shaw, and William J. Murphy (National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Cincinnati, OH 45226, ckardous@cdc.gov)

The National Institute for Occupational Safety and Health (NIOSH) conducted a follow-up study to examine the accuracy of smartphone sound measurement applications (*apps*) using external, calibrated, microphones. In the initial study, we examined 192 *apps* on the Apple (iOS) and Google (Android) platforms. Overall, 10 iOS *apps* met our selection criteria for occupational noise exposure measurements, and of those, only 4 iOS *apps* (SoundMeter, SPLnFFT, SPL Pro, and NoiSee) were within ± 2 dB(A) of a reference microphone. For this study, we selected the same 4 iOS *apps* and examined their accuracy and performance using the MicW i436 and the Dayton Audio iMM-6 external microphones. The MicW i436 microphone is marketed as compliant with IEC-61672 class 2 sound level meter standard. Testing was conducted in a reverberant chamber using pink noise over seven nominal sound levels from 65 to 95 dB. The results showed an even closer agreement than the earlier study, with mean differences within ± 1 dB(A) of the reference microphone. This study suggests that the use of external microphones, and the ability to calibrate the microphones and *apps*, improves the overall accuracy and precision of measurements, and removes many of the constraints and limitations associated with the built-in smartphones' microphones.

8:25

2aNSa2. Investigating the feasibility of using mobile devices for remote noise monitoring and data acquisition. Mike Dickerson (MD Acoust., 4960 S. Gilbert Rd., Ste 1-461, Chandler, AZ 85249, mike@mdacoustics.com)

The feasibility of using the Faber Acoustical SoundMeter Pro app for mobile devices for remote noise monitoring and data acquisition was evaluated and tested. A data file acquisition protocol was created, a remote database storage system was developed, and a reporting system was prepared to test the feasibility. The data and results of this assessment are presented and discussed.

8:45

2aNSa3. Assessment of noise exposures for pre-term infants during air transport to neonatal intensive care units using iPhone sound meter apps. William W. Clark (Audiol. Comm Sci., Washington Univ. School of Med, 660 S. Euclid Ave., Box 8042, St. Louis, MO 63110, clarkw@wustl.edu) and Scott Saunders (Pediatrics/Newborn Medicine, Washington Univ. School of Med., St. Louis, MO)

A significant number of infants born prematurely or with life-threatening conditions in local hospitals require transport to a regional tertiary care center. St. Louis Children's Hospital's (SLCH) neonatal intensive care unit (NICU) serves southern Missouri and Illinois, and 3000 pre-term neonates are transported to the hospital annually, most by helicopter or fixed wing aircraft. This initial study evaluates the accuracy and efficacy of using an iPhone app (SoundMeterPro,Faber Acoustics) for routine collection of infants' noise exposures inside the isolette during air transport. The app was downloaded onto iPhones (4,6S) calibrated in a sound field using a Larson-Davis type I sound level meter (831). The meter and the iPhones were placed inside an unoccupied isolette and recorded in-flight noise levels during the outbound portion of trips from the NICU to the local hospital. Sound levels were high (85–90 dBA) and higher during take-off and landings or when the isolette lid was opened, and level and OBN measures obtained with the SoundMeterPro app were similar to those from the type 1 SLM. An additional advantage of the iPhone SLM platform for routine use in aircraft is that it does not require FAA certification for each aircraft, if used in the airplane mode.

9:05

2aNSa4. 2015 Jacksonville Executive at Craig Airport noise study. Herbert R. Matthews (Aviation, Jacksonville Univ., 1806 Pleasant Point Ln., Jacksonville, GA 32225, bmatthe2@jacksonville.edu) and Ross Stephenson (Aviation, Jacksonville Univ., Jacksonville, FL)

This research will update outdated information as identified in the original 2006 Federal Aviation Regulation (FAR) Part 150 Study of Craig Municipal Airport. The original study conducted an evaluation of the Airport's existing noise conditions from 2004 to 2005 to determine if current voluntary operational procedures were achieving their desired effect, and identified other opportunities to reduce aircraft-related noise impacts on the communities surrounding the airport. This study will examine and revalidate the findings of the original 2006 study. It is expected that the noise level contours, as depicted within the original study, have reduced in size since 2006. This is due to the utilization of more technically advanced aircraft engines on today's modern business jets. Airport noise magnitude, frequency, and duration will be measured during daily operations. Sound exposure level (SEL) metric used by the Federal Aviation Administrations will be used to measure single event noise exposure levels along with time weighted cumulative noise metrics. The findings of the study may provide the opportunity for the Jacksonville Airport Authority (JAA), the aviation industry, affected political jurisdictions and airport neighbors to work together in the evaluation of potential noise reduction and land use control measures.

9:25

2aNSa5. Real-time sound measurements of exercise classes with mobile app demonstrate excessive noise exposure. Sumi Sinha, Elliott D. Kozin, Matthew R. Naunheim, Samuel R. Barber, Kevin Wong (Otolaryngol., Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA 02114, sumi_sinha@meei.harvard.edu), Leanna W. Katz (Spaulding Rehabilitation Hospital, Boston, MA), Ishmael J. Stefanov-Wagner, and Aaron K. Remenschneider (Otolaryngol., Massachusetts Eye and Ear Infirmary, Boston, MA)

Noise induced hearing loss is a major contributor to observed hearing impairment in the general population. Advanced mobile technology now allows the opportunity to directly measure noise exposure profiles on an individual basis. Herein, we conduct a pilot study to determine noise exposure in popular gym-based exercise cycling classes where music and equipment generate high intensity sound. A calibrated iOS mobile app (SoundMeter, Faber Acoustical) coupled to a microphone was used to measure noise in a random sampling of similar indoor cycling classes ($n = 7$). The average length of exposure was 52.33 ± 3.81 min. Maximum sound levels recorded across all classes were 125.96 dB and averaged 119.6 ± 3.5 dBA. By NIOSH standards, 31.5 ± 14.7 min were spent over 100 dBA, corresponding to a class daily exposure dose of $191\% \pm 1.2\%$. The 8-h projected dose was $1781\% \pm 11.9\%$. Preliminary data suggest that certain exercise classes may expose participants to excessive, potentially dangerous sound levels. The use of readily accessible mobile technology can assess noise exposure risk from the consumers' perspective, allowing a new level of independent self-monitoring of noise and empowering individuals to become active participants in their hearing health.

Contributed Papers

9:45

2aNSa6. Noise illustrated. Michael Ermann (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu)

Principles and research findings on the subject of noise control are presented in the graphic language of architecture in the hope that not only architects, but others will benefit from the translation. (1) Mass, airtightness, and structural discontinuity are illustrated with baseline/better comparisons; (2) transmission loss data are culled and translated from tables to graphs for easy contrasts, (3) families of assemblies are illustrated and ranked by STC and IIC values, (4) the frequency domain of impact noise is explored graphically, (5) barrier design principles are demonstrated, (6) AHU fan noise and turbulence attenuation strategies are presented visually, and (7) common in-the-field construction mistakes are drawn to illustrate flanking paths, vibration isolation short-circuiting, and barrier construction acoustic bridging. Rules-of-thumb and nomographs are offered for early-design best-practices in building site selection, duct distances, ducted air velocities, window selection, and space planning. The illustrations presented come from over 250 drawings in the book, *Architectural Acoustics Illustrated* (Wiley, 2015), half of which is dedicated to noise control.

10:00

2aNSa7. A comparison of readily available sound level apps with respect to field performance and applications in architectural education. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com) and Emily McGlohn (School of Architecture, MS State Univ., MS State, MS)

This investigation was driven first by a suggestion that apps be used by residents, sound creators, and police to evaluate sound sources in regard to regulations, and then further in an effort to understand how apps might be useful for architecture students studying active building or environmental systems. To simulate a typical educational or "street" situation, soundwalks are taken in which various sources are measured using apps on different phones as well as a Type 1 sound level meter for reference. The results are compared and a recommendation of appropriate application is made.

Session 2aNSb

Noise, Structural Acoustics and Vibration, and Architectural Acoustics: Noise and Vibration Impacts from Crossfit Training Facilities

Steve Pettyjohn, Cochair
The Acoustics & Vibration Group, Inc., 5765 9th Avenue, Sacramento, CA

James E. Phillips, Cochair
Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Chair's Introduction—10:40

Invited Papers

10:45

2aNSb1. The effectiveness of resilient sports flooring on noise from crossfit activities. Matthew V. Golden and Paul Gartenburg (Pliteq, 1370 Don Mills Rd., Unit 300, Toronto, ON M3B 3N7, Canada, mgolden@pliteq.com)

Crossfit facilities in dense, multi-use facilities are growing in numbers every day. Even minimally sensitive receivers can be very significantly impacted by Crossfit activities. Consequently, vibration and vibration induced noise in structures as a result of heavy weight drops are a never ending issue. This paper compares data recorded from heavy weight drops in an ASTM E492 floor/ceiling testing suite. The resultant sound pressure level in the receiving room and vibration levels in the structure are measured. Several different standard and high performance resilient sports floorings are compared. These sports floors are tested on single concrete slabs as well as various floating slabs. The floating slabs include continuous rubber underlayments, discrete rubber isolators and spring jack-up floating floors. In addition, laboratory impact and airborne isolation performance results of a spring jack-up floating floor are reported.

11:05

2aNSb2. Mock-up testing for CrossFit vibration. David Manley and Ben Bridgewater (D.L. Adams Assoc., Inc., 1536 Ogden St., Denver, CO 80218, dmanley@dlaa.com)

A new construction mixed-use building was proposed in Crested Butte, CO, to consist of ground floor fitness, second floor commercial office, and third floor residential. It was anticipated that the ground floor fitness space would include CrossFit style workouts which often incorporate Olympic weight lifting and dropping activities. Due to the remote project location, five different mock-ups of impact and vibration isolation products were assembled in a gym in Denver to test the reduction in vibration levels from heavy weight impacts. Acceleration levels were measured on the concrete slab on grade close to the weight drop and in the soil outside the gym area. Mock-up testing methodology and results are presented. Based on the test results, recommendations on products, assemblies, and construction details were provided.

11:25

2aNSb3. Impact of heavy weight drops at Crossfit® training facilities on adjacent doctor's examination rooms and on conference rooms. Steve Pettyjohn (The Acoust. & Vib. Group, Inc., 5765 9th Ave., Sacramento, CA, spettyjohn@acousticsandvibration.com)

Significant sound and vibration are generated when members of Crossfit® training facilities drop heavy barbell weights to the floor from waist or shoulder height. Weights between 40 kg (90 lbs) and 100 kg (225 lbs) are repeatedly lifted from the floor to should height, then dropped to the floor and then picked up again. Crossfit® combines a variety of exercises using weights, pull up bars, and tires for dragging to increase the strength of participants. The sound and vibration generated by dropping of the weights is the only activity that has generated strong reaction from adjacent spaces. The Crossfit® facilities are found in many spaces, but often in warehouse/office or strip offices, i.e., locations with many types of tenants. Examples are Crossfit® facilities next to optometrists, conference rooms, and massage parlors. The sound and vibration generated by the weight drops and resulting sound and vibration in the receiver space are presented for several conditions of the equipment and the floor. Several options for reducing the transmission of excessive sound and vibration into adjacent spaces is presented. These include modifications to the weights, floor toppings, floor dimensions, and wall construction.

Contributed Paper

11:45

2aNSb4. Investigation of acoustical environment of multi-story buildings in Korea for applicability of EN 12354. Sung M. Kim, Hansol Lim, and Jin Y. Jeon (Architectural Eng., Hanyang Univ., Seongdong-gu Wangsimni-ro 222, Seoul 133-791, South Korea, rainbear0622@gmail.com)

In multi-story buildings, sound reduction can be evaluated using sound prediction model defined in the EN 12354 in Europe generally. The reduction model deals with the calculation and evaluation of direct and flanking transmission between two rooms through each structure elements for example the floor, ceiling, inner wall, or etc. In case of Korea, general residential type of is multi-story building so the noise transmission between neighbors is the most well-known acoustical problem. The EN 12354 can be considered as an important evaluation method, however the acoustic environment in Korea should be considered for the applicability of prediction model. Therefore, acoustical environments of multi-story buildings were investigated using field survey in order to evaluate the applicability through EN 12354.

TUESDAY MORNING, 24 MAY 2016 SALON H, 8:00 A.M. TO 11:45 A.M.

Session 2aPA

Physical Acoustics and Biomedical Acoustics: Vortex Beams and Radiation Torque Physics I

Philip L. Marston, Cochair
Physics and Astronomy Dept., Washington State University, Pullman, WA 99164-2814

Likun Zhang, Cochair
University of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199

Chair's Introduction—8:00

Invited Papers

8:05

2aPA1. Radiation stress, momentum, angular momentum, and power relationships for axisymmetric objects in acoustic vortex and coaxial wavefields. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu) and Likun Zhang (Phys. Dept. & Ctr. for Nonlinear Dynam., Univ. of Texas at Austin, Austin, TX)

Starting with P. Westervelt's formulation of the radiation stress tensor and G. Maidanik's application to angular momentum in the 1950s, some recent theorems concerning acoustic radiation forces, torques, angular momentum, and power flow will be illustrated. Applications include acoustic vortex wavefields, Bessel beams, and other invariant beams. While derivations are simplified for objects surrounded by inviscid fluids [L. Zhang and P. L. Marston, Phys. Rev. E. **84**, 035601 (2011); L. Zhang and P. L. Marston, Phys. Rev. E. **84**, 065601 (2011); L. Zhang and P. L. Marston, J. Acoust. Soc. Am. **131**, EL329–EL335 (2012)], for spheres in slightly viscous liquids limiting approximations are known at long wavelengths that recover results derivable by other approaches [L. Zhang and P. L. Marston, J. Acoust. Soc. Am. **136**, 2917–2921 (2014); P. L. Marston, POMA **19**, 045005 (2013)]. Theorems obtained are helpful for relating negative radiation forces to the asymmetry of the scattering pattern for spheres in Bessel beams. The relation with extinction theorems is noted [L. Zhang and P. L. Marston, Bio. Opt. Express **4**, 1610–1617 (2013); (E) **4**, 2988 (2013)] as well as momentum and angular momentum radiated by sources. [Work supported in part by ONR.]

8:30

2aPA2. Wave vortices, structured beams, and their interaction with matter: What can we learn from analogies? Karen Volke-Sepulveda (Instituto de Fisica, Universidad Nacional Autonoma de Mexico, Apdo. Postal 20-364, Mexico City, Mexico D.F. 01000, Mexico, karen@fisica.unam.mx)

One of the many reasons making the study of waves so fascinating is the opportunity it brings to establish links among different areas of physics. In particular, the analogies between optics and acoustics have led to important advances in both directions. Fundamental research on structured wave fields and phase singularities, such as Bessel beams and helical waves, as well as their interaction with matter, is one of the areas that has been benefited greatly from these cooperative advances. Regarding topological studies of scalar wave fields, for instance, sound waves might represent a more versatile alternative than light, since their phase structure can be measured directly with respect to a reference signal. In contrast, the intensity distribution of light can be directly observed, whereas the intensity

distribution of an acoustic field has to be determined from measurements of the sound pressure. Therefore, optical and acoustical demonstrations complement and reinforce each other. In this presentation, some specific examples of this productive interchange will be discussed as a starting point, involving the study of acoustic vortices. This will pave the way for introducing some recent results on the analysis of structured light beams, which may find direct translation to the acoustic realm.

8:55

2aPA3. Angular momentum conservation and symmetry. Likun Zhang (Univ. of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199, lzhang@chaos.utexas.edu), Yong Li (CNRS-Université de Lorraine, Vandœuvre-lés-Nancy, France), Xue Jiang, Bin Liang, and Jian-chun Cheng (Nanjing Univ., Nanjing, China)

The conservation of angular momentum during the interactions of sound fields with objects is associated with the symmetry in the fields and the objects. Simple arguments of symmetry and/or conservation can have efficient applications in analyses of angular momentum transfer during the interactions. Scenarios that will be illustrated include: (a) conserved phase distribution for analysis of torques exerted by vortex beams on axis-symmetric objects for its connection with absorption [L. Zhang and P. L. Marston, JASA **129**, 1679–1680 (2011); Phys. Rev. E. **84**, 065601 (2011)], (b) symmetric superposition of vortex beams for analysis of torques exerted by orthogonal waves on a small compressible object in a slightly viscous fluid [L. Zhang and P. L. Marston, JASA **136**, 2917–2921 (2014)], and (c) broken symmetry in a planar meta-screen for introducing angular momentum into a plane sound field of null angular momentum [Y. Li *et al*., Phys. Rev. Appl. **4**, 024003 (2015); X. Jiang *et al*., in preparation].

Contributed Paper

9:20

2aPA4. Vortex beams and radiation torque for kidney stone management. Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Michael Bailey, Bryan W. Cunitz (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, mike.bailey.apl@gmail.com), M. Terzi (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Seattle, Washington), A. Nikolaeva, S. Tsysar, and Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

Our team previously developed an instrument to reposition kidney stones with acoustic radiation force. In a clinical trial, the technology was used to transcutaneously facilitate passage of small stones and to relieve pain by dislodging obstructing large stones. Acoustic trapping and manipulation of kidney stones in water has recently been investigated using both single element and sector arrays in the range of 0.3–1.5 MHz. Experimental holographic reconstruction of the transducer surface velocity confirmed the proper operation of each transducer. Human stones approximately 5 mm, as well as glass and aluminum beads, were placed on a flat tissue phantom in a water bath. During exposure, stones were drawn to the beam axis, and then controllably translated along the surface in any direction transverse to the beam. The phase between sector elements could be used to control the vortex size, as well as rate and direction of rotation of the trapped object. The trapping effect was disrupted at increased transducer output, possibly by generation of acoustic streaming. In conclusion, a method was tested for transverse acoustic trapping of kidney stones with vortex beams. [This work was supported by RBBR 14-02-00426, NIH NIDDK DK43881, DK104854, and DK092197, and NSBRI through NASA NCC 9-58.]

Invited Papers

9:35

2aPA5. Characterization of phased array-steered acoustic vortex beams. Jhon F. Pazos-Ospina (School of Mech. Eng., Universidad del Valle, Cali, Valle del Cauca, Colombia), Ediguer E. Franco (Universidad Autónoma de Occidente, Cali, Colombia), and Joao L. Ealo (School of Mech. Eng., Universidad del Valle, Acoust. and Vib. Lab., Ciudad Universitaria Meléndez. Bldg. 351., Cali, Valle del Cauca 760032, Colombia, joao.ealo@correounivalle.edu.co)

Acoustic vortex (AV) beams generation is a subject of current interest. Even though different applications have been proposed using AV, their potential of use is still to be explored. Recent research works on particle manipulation use phased array systems for AV generation because it allows a flexible beam configuration, i.e., the beam can be easily focalized and modified in its shape. However, little attention has been paid to the fact that the AV can also be electronically steered. In view of this, this work presents an study of the steering capability of an AV. In particular, we analyze the effect of the applied delay law on the structure of AV beams steered at different angles using an array transducer of 32 equidistant elements, deployed on a triangular lattice, operating at 40 kHz. Special attention is paid to the appearance of grating vortices and their characteristics. The effect of the individual element directivity on the resultant beam is also studied. Experimental measurements were carried out in order to validate numerical estimations. Obtained results paves the way for the use of electronically steered vortices in different applications. Also, the potential of use of acoustic grating vortices is discussed.

10:00–10:15 Break

10:15

2aPA6. Sonic Screwdrivers and Tractor Beams: what the Acoustics Community often gets *wrong* about gradient forces versus radiation pressure. Gabriel C. Spalding (Phys., Illinois Wesleyan Univ., 201 E. Beecher St., Bloomington, IL 61701-7222, gspaldin@iwu.edu), Patrick Dahl (Phys., Illinois Wesleyan Univ., Evanston, IL), Zhengyi Yang (Inst. for Medical Sci. & Technol., Univ. of Dundee, St. Andrews, United Kingdom), Peter Glynne-Jones (Eng. Sci., Univ. of Southampton, Southampton, United Kingdom), Michael P. MacDonald (Inst. for Medical Sci. & Technol., Univ. of Dundee, Dundee, United Kingdom), Christine Demore, and Sandy Cochran (School of Eng., Univ. of Glasgow, Glasgow, United Kingdom)

Through discussion of our sonic screwdriver and tractor beam experiments, we aim to highlight the respective advantages of conservative and non-conservative forces. Commonly in acoustic trapping, conservative, gradient-induced mechanisms (e.g., standing waves) are used to manipulate matter. Such situations are reasonably described in terms of potential energy landscapes, an approach also applied to optics, for applications such as cell sorting [MacDonald et al, Nature 426 (2003)]. No such description is possible for radiation pressure, which is non-conservative, a distinction that is sometimes muddled in the literature, although it was made clear even in early work [e.g., King, Proc. R. Soc. Lond. A **147** (1934); Gor'kov, Sov. Phys. Doklady **6**, 773 (1962)]. Our "sonic screwdriver" makes use of two non-conservative mechanisms: levitation by radiation pressure and rotation by transfer of azimuthal momentum components [Demore *et al*., PRL **108** (2012)]. We also note that the term "tractor beam" has often been reserved to describe an effect involving non-conservative forces, and demonstrate an attractive force produced in such an arrangement, even against a net momentum flux [Demore *et al*., PRL **112** (2014)].

10:40

2aPA7. Single-beam acoustical tweezers: Concept, theory, and experimental realization. Diego Baresch (Institut des NanoSci. de Paris, UPMC, 4, Pl. Jussieu, Paris 75005, France, diego.baresch@upmc.fr), Régis Marchiano (Institut Jean Le Rond D'Alembert, UPMC, Paris, France), and Jean-Louis Thomas (Institut des NanoSci. de Paris, UPMC, Paris, France)

Vortex beams are characterized by wavefronts twisting around their axis of propagation. A three dimensional region of silence tightly bounded by a high intensity ring emerges from the phase's screw dislocation, which can be used for particle manipulation. In this paper, we present the first experimental realization of selective 3D acoustical tweezers with a single beam. It had long been recognized that acoustic radiation pressure pushed objects in the direction of propagation. As we will show, by accurately modeling the radiation pressure exerted on an elastic sphere lying in any arbitrary location, it is possible to identify this axial ejection and to propose a solution to counterbalance this effect. The acoustic puling force of a properly focused vortex beam is explained as a result of the particle's monopolar mode annihilation on the propagation axis. Consequently, these beams act as a stable trap in three dimensions. An experimental realization has been proposed in water using a ultrasonic beam (1 MHz). Elastic particles in the sub-millimetric range have been accurately trapped and manipulated. This setup's dexterity and selectivity is the macroscopic analogous of optical tweezers exerting, however, forces which are 5 orders of magnitude stronger at comparable intensities.

11:05

2aPA8. Taming tornadoes: Controlling orbits inside acoustic vortex traps. Asier Marzo, Mihai Caleap, and Bruce W. Drinkwater (Mech. Eng., Univ. of Bristol, University Walk, Queen's Bldg. 1.1, Bristol, Bristol BS8 1TR, United Kingdom, amarzo@hotmail.com)

Bessel-shaped acoustic vortices have been theorized to be ideal candidates for tractor beams—that is, acoustic beams that exert forces opposite to the direction of propagation. Recently, it has been shown experimentally that these pulling forces exist and that particles can be trapped inside a focused vortex beam, both in water and air. However, our experiments and simulations reveal that the behavior of particles trapped inside a focused vortex is strongly dependent on the particle size. We trapped expanded polystyrene spheres (ranging from 0.6 mm to 4 mm diameter) 4 cm above a phased array made of 52 transducers (40 kHz and 1 cm diameter each) that generated an airborne acoustic vortex. Depending on its size, a particle remained trapped, oscillated between being trapped and orbiting, orbited or was ejected; only small particles (< 1 mm) were stably trapped. We then devised a new procedure for making vortex traps stable for a wider range of particle sizes; the strategy relies on switching the direction of the vortex at 4 kHz. In addition, by regulating the amount of time that each direction is emitted, it is possible to controllably rotate symmetric particles. This enables new applications in particle manipulation such as centrifugation of cells or actuation of micromachines.

Contributed Paper

11:30

2aPA9. Helicity sensitive processing and imaging of a first order vortex beam scattered by a sphere. Viktor Bollen (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, viktor.bollen@wsu.edu), Daniel S. Plotnick (Acoust., Appl. Res. Lab, Seattle, WA), and Philip L. Marston (Washington State Univ., Pullman, WA)

A vortex beam's wavefield has a null on the axis of propagation and an angular phase ramp proportional to the order of the beam. The rapid phase ramp at the null leads to the possibility of high-resolution imaging and precise alignment of the system. Using a modified four-panel transducer, a first order vortex beam was generated by driving each panel with an appropriate phase shift [V. Bollen *et al*., Proc. Meet. Acoust. **19**, 070075 (2013)]. Utilizing this transducer, a solid sphere was insonified and the backscattering measured. Recording the backscattering on each panel separately allowed selection between helicity neutral and helicity sensitive detection modes, without changing the experimental setup, by introducing individual phase shifts in post-processing [V. Bollen, *et al*., J. Acoust. Soc. Am. **137**, 2439 (2015)]. Using time delay-and-sum imaging algorithms, we created high-resolution three-dimensional profiles of the beam, relating the sphere location to the beam pattern. With cross-correlation involving measured and computed scattering amplitudes, we showed agreement with a Kirchhoff integration based simulation of the beam. [Work supported by ONR.]

Session 2aPP

Psychological and Physiological Acoustics and Signal Processing in Acoustics: Approaches to Improve Speech Understanding in Noise

Eric Healy, Cochair
Speech & Hearing Science, The Ohio State University, Pressey Hall Rm. 110, 1070 Carmack Rd., Columbus, OH 43210

Ying-Yee Kong, Cochair
Speech Language Pathology & Audiology, Northeastern University, 226 Forsyth Building, 360 Huntington Ave., Boston, MA 02115

Tao Zhang, Cochair
Signal Processing Reseearch, Starkey Hearing Technologies, 8602 Zachman Circle, Eden Prairie, MN 55344

Chair's Introduction—8:00

Invited Papers

8:05

2aPP1. Understanding the speech-understanding problems of older adults. Larry E. Humes (Dept Speech & Hearing Sci., Indiana Univ., Bloomington, IN 47405-7002, humes@indiana.edu)

This presentation will provide an overview of the nature of the difficulties experienced by older adults with impaired hearing when listening to speech in noise. Many older adults may experience a "triple whammy" of difficulties: (1) peripheral pathology in the cochlea or auditory nerve; (2) age-related deficits in central-auditory function; and (3) age-related changes in cognition that impact speech perception. Each of these difficulties may appear in isolation or in various combinations for a given older adult. This presentation will review the evidence surrounding each of these potential sources of difficulty and the relative contribution of each across various types of "noise" backgrounds. In addition, the relative importance of each source of difficulty for aided and unaided listening conditions will be considered. [Work supported, in part, by NIA R01 AG008293.]

8:25

2aPP2. Interactions between spectral and temporal processing on speech understanding in cochlear-implant users. Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Understanding speech in noise remains one of the greatest challenges facing people with cochlear implants (CIs). For normal-hearing listeners, speech understanding in noise seems limited to a large extent by the modulation energy (temporal fluctuations) present in the noise, rather than the noise energy itself. This finding does not seem to apply in CI users, for whom no difference is observed between maskers with or without inherent temporal fluctuations. The current explanation for this qualitative difference between the results from normal-hearing listeners and CI users is based on the lack of spectral resolution in CIs, which results in an overlap of stimulation from adjacent channels, and a resultant smoothing of the temporal envelopes produced by noise. This study reviews evidence for this hypothesis, and tests some arising predictions, including the effects of narrower stimulation patterns (tripolar versus monopolar) and increased spacing between active electrodes in both CI users and normal-hearing listeners using envelope-vocoder simulations of CI processing. [Work supported by NIH grant R01 DC012262.]

8:45

2aPP3. Efficacy and viability of an algorithm to improve speech understanding in noise for the hearing impaired. Eric W. Healy (Speech & Hearing Sci., The Ohio State Univ., Pressey Hall Rm. 110, 1070 Carmack Rd., Columbus, OH 43210, healy.66@osu.edu)

A primary complaint of hearing-impaired individuals involves poor speech understanding when background noise is present. Hearing aids and cochlear implants often allow good speech understanding in quiet backgrounds. However, the listeners' noise intolerance and the devices' inability to effectively combat background noise often conspire to produce poor performance in noise. Perhaps surprisingly, effective solutions to this problem have remained elusive despite considerable effort. One promising solution involves a single-microphone algorithm to extract speech from background noise. The algorithm is based on the concept of the ideal binary or ratio mask, and employs standard machine-learning techniques to train a deep neural network to estimate the mask, given only the speech-plus-noise mixture. Existing data indicate that large intelligibility increases by hearing-impaired listeners may be obtained across a variety of noisy conditions. In this talk, an overview of this approach will be provided, and the potential for implementation into hearing aids and cochlear implants will be discussed. [Work supported by NIH.]

9:05

2aPP4. Enhancement of spectral contrast and spectral changes as approaches to improving the intelligibility of speech in background sounds. Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

Cochlear hearing loss is accompanied by reduced frequency selectivity, which contributes to problems in understanding speech in background sounds. The excitation pattern evoked by sounds like vowels is "flatter" in an impaired ear than in a normal ear. Two approaches to partially compensating for the effects of reduced frequency selectivity are described. The first uses spectral contrast enhancement on a frame-by-frame basis. Only features of the short-term spectrum that would be represented in a normal ear are enhanced. Experimental evaluations of this approach showed that it could improve the intelligibility of speech in noise by a small amount, but that this required a fine frequency analysis, in turn requiring relatively long frames and time delays that might be too long for use of the method in hearing aids. A second approach enhances spectral changes over time, based on the idea that information is speech is carried by spectral changes. This approach also led to modest improvements in the intelligibility of speech in noise, especially when the amount of enhancement was made to vary with frequency depending on the hearing loss at that frequency and when the parameters of the processing were selected for each individual using a genetic algorithm.

9:25

2aPP5. Metrics for evaluating noise suppression in hearing aids. James Kates (Speech Lang. Hearing Sci., Univ. of Colorado., 409 UCB, Boulder, CO 80309, James.Kates@colorado.edu)

Noise suppression in audio systems such as hearing aids typically involves modifying the speech envelope. Noisy speech segments having poor signal-to-noise ratios are attenuated while those considered to be primarily speech are left at or near full intensity. Both spectral subtraction and ideal binary mask noise suppression work on this principle. Recent work in the development of speech intelligibility and speech quality metrics indicates that the envelope fidelity, computed as the correlation coefficient between the processed noisy speech envelope and the original noise-free speech envelope, is an important factor in determining noise-suppression benefit. This result implies that the best noise suppression system is one in which the envelope of the processed noisy speech is restored to match as closely as possible that of the original noise-free speech. The potential benefits and limitations of envelope restoration for noisy speech are analyzed using the HASPI intelligibility and HASQI quality metrics, and the performance of existing noise suppression processing is compared to that of envelope restoration for both normal-hearing and hearing-impaired listeners.

9:45–10:00 Break

10:00

2aPP6. A novel binaural speech enhancement algorithm for hearing impaired listeners. Tao Zhang (Signal Processing Res., Starkey Hearing Technologies, 8602 Zachman Circle, Eden Prairie, MN 55344, tzhang28@ieee.org)

With the advent of wireless ear to ear communication, binaural speech enhancement becomes feasible in commercial products such as hearing aids. The first such an example is a fixed binaural beam former that attenuates noise from the nonlook direction while preserving the speech in the look direction. While such an algorithm has shown significant benefits in the lab, it has not produced the wow effect in the field as we have all expected. In this paper, we will discuss some of the practical issues limiting the benefits of such an algorithm in the field. In addition, a novel binaural speech enhancement algorithm is proposed by balancing robustness along with optimal performance. Its benefits will be demonstrated by comparing it with the leading binaural speech enhancement algorithms. Furthermore, future research directions will be discussed.

10:20

2aPP7. Comprehensive evaluation of binaural hearing aid pre-processing strategies—Speech intelligibility in realistic noise scenarios. Stephan Ernst, Regina Baumgärtel, Christoph Völker, Anna Warzybok, Mathias Dietz, Volker Hohmann, and Birger Kollmeier (Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, Carl von Ossietzky Universität Oldenburg, Oldenburg 26111, Germany, stephan.ernst2@uni-oldenburg.de)

For the development of new signal processing approaches, e.g., for hearing aids, a final stage of evaluation is crucial. However, the evaluation procedures for assessing performance and benefit often represent a vast and not unified realm. Thus, we suggest a comprehensive evaluation to describe the efficacy, i.e., the anticipated real-world benefit as precise as possible. This comprises physical, instrumental, and perceptual (human) measurements of objective measures, as well as subjective attributes. Eight signal pre-processing strategies were evaluated following this concept, including directional microphones, coherence filters, single-channel noise reduction, binaural beamformers, and their combinations. Speech reception thresholds (SRTs) were measured with normal-hearing and hearing-impaired listeners in three realistic noise scenarios and compared with predictions of common instrumental measures. Although hearing-impaired listeners required a better signal-to-noise ratio to obtain 50% intelligibility than listeners with normal hearing, no differences in SRT benefit (of up to 4.8 dB) from the different algorithms were found between the two groups. This suggests a possible application of noise reduction schemes for listeners with different hearing status. Although the instrumental measures can predict the individual SRTs without pre-processing, development is necessary to predict the benefits obtained from the algorithms at an individual level.

10:40

2aPP8. How individual differences in sensory coding and attentional control impact understanding speech in noise. Barbara Shinn-Cunningham (Biomedical Eng., Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu), Dorea Ruggles (Psych., Univ. of Minnesota, Minneapolis, MN), Inyong Choi (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA), Hari Bharadwaj (Athinoula A. Martinos Ctr. for Biomedical Imaging, Massachusetts General Hospital, Boston, MA), Golbarg Mehraei (Elec. Eng., Tech. Univ. of Denmark, Lyngby, Denmark), and Lengshi Dai (Biomedical Eng., Boston Univ., Boston, MA)

Historically, the majority of psychoacoustic studies of hearing ability have viewed individual differences as noise: a nuisance that makes it difficult to see the effects that different acoustic conditions have on auditory perception. This talk reviews how we have begun to use individual differences to tease apart the processes that affect perception, with a particular focus on how listeners understand speech when there are competing sound sources. We find that individual subjects show consistent differences in their ability to understand speech in noise. These consistent differences can come both from differences in the fidelity of sensory coding and from differences in the ability to focus selectively on important sound and suppress unimportant sound. Importantly, which of these factors predicts performance depends greatly on the details of the stimuli used in a given task, and what stage of processing is the resulting bottleneck, determining performance. When fine differences in the sound content, such as small differences in location or pitch, are critical for a task, differences in sensory fidelity dominate individual differences in ability. However, when the acoustic features important for separating sound streams and identifying the target stream from the mixture are very distinct, individual differences in ability reflect differences in attentional control. These results highlight how understanding speech in noise depends on complex interactions between the ear and the brain.

11:00

2aPP9. The "meaning" in noise—Evidence for bottom-up information masking of within channel modulation coding of speech. Simon Carlile (Starkey Hearing Res. Ctr., F13, Anderson Stuart Bldg., Camperdown, Sydney, New South Wales 2006, Australia, simonc@physiol.usyd.edu.au)

Spectro-temporal variations are a consequence of dynamic structural variations of a sounding body. The human auditory system is highly optimized for the detection, segregation and analysis of one class of such variations—speech. Here, we will examine some consequences of this optimisation in the context of complex listening involving multiple concurrent talkers. In such conversational settings, listeners rapidly shift their attention from one to another talker so foreground and background are defined dynamically by listener intent. Up-regulation of the attended-to talker is generally thought to result from endogenous, top-down attention. Here, we review a recent report indicating that substantial informational masking between concurrent talkers may result from bottom-up interactions between sources within frequency modulation channels. This indicates that temporally dynamic aspects of within-frequency channel processing plays a very important role in speech masking by the sort of "noise" most commonly encountered in natural conversational settings. Possibly even more surprising, these interactions appear to be modulated by spatial attention. Attentional enhancement of a foreground sound might then also involve processes at the level of the within-frequency channel. Whether this occurs prior to or as a consequence of grouping is a question of some functional significance but also points to the importance of knowing the listeners focus of attention.

11:20

2aPP10. Neural representations of speech, and speech in noise, in human auditory cortex. Jonathan Z. Simon (Dept. of Elec. & Comput. Eng., College Park, MD 20742, jzsimon@umd.edu)

We investigate how continuous speech, whether presented alone, degraded with noise, or masked by other speech signals, is represented in human auditory cortex. We use magnetoencephalography (MEG) to record the neural responses of listeners to continuous speech in a variety of contexts. We find that cortical representations of continuous speech are robust to noise under a wide variety of conditions, including clean speech, additive stationary noise, band-vocoded speech and noise, and speech with competing speakers. In the last case, individual neural representations of the speech of both the foreground and background speaker are observed, with each being selectively phase locked to the rhythm of the corresponding speech stream. In all observed cases, the temporal envelope of the acoustic speech stream can be reconstructed from the observed neural response to the speech.

11:40

2aPP11. Neuro-steered noise suppression for auditory prostheses. Tom Francart, Neetha Das (NeuroSci., KU Leuven, Herestraat 49 Bus 721, Leuven 3000, Belgium, tom.francart@med.kuleuven.be), Simon Van Eyndhoven, Wouter Biesmans, and Alexander Bertrand (Elec. Eng., KU Leuven, Leuven, Belgium)

In a multi-speaker scenario, a major challenge for noise suppression systems in hearing instruments is to determine which sound source the listener is attending to. It has been shown that a linear decoder can extract a neural signal from EEG recordings that is better correlated with the envelope of the attended speech signal than with the envelopes of the other signals. This can be exploited to perform auditory attention detection (AAD), which can then steer a noise suppression algorithm. The speech signal is passed through a model of the auditory periphery before extracting its envelope. We compared 7 different periphery models and found that best AAD performance was obtained with a gamma-tone filter bank followed by power-law compression. Most AAD studies so far have employed a dichotic paradigm, wherein each ear receives a separate speech stream. We compared this to a more realistic setup where speech was simulated to originate form two different spatial locations, and found that although listening conditions were harder, AAD performance was better than for the dichotic setup. Finally, we designed a neuro-steered denoising algorithm that informs the voice activity detection stage of a multi-channel Wiener filter based on AAD, and found a large signal-to-noise-ratio improvement at the output.

Session 2aSC

Speech Communication: Intelligibility, Hearing Impairment, Aging (Poster Session)

Ewa Jacewicz, Chair
Department and Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey Hall, Columbus, OH 43210

All posters will be on display from 8:00 a.m. to 11:30 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 9:30 a.m., and authors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m.

Contributed Papers

2aSC1. Band importance functions of listeners with cochlear implants. Adam K. Bosen and Monita Chatterjee (Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, adam.bosen@boystown.org)

Band Importance Functions (BIFs) are measured by dividing the acoustic spectrum into discrete bands and estimating the relative contribution of each band to speech intelligibility. Previous studies have demonstrated strong across-subject consistency for normal hearing (NH) listener BIFs. However, it is unlikely that cochlear implant (CI) listeners will demonstrate similar consistency, because many physiological and psychophysical properties of hearing vary across CI listeners and across electrodes within an ear. Here, we measured BIFs by testing speech intelligibility of IEEE sentences filtered to contain a random subset of bands from a listener's clinical MAP and regressing across trials to determine each band's average contribution. BIFs obtained from NH listeners with vocoded speech that either did or did not simulate cross-band interaction followed the characteristic inverted "U" shape with a peak around 1–2 kHz that has been previously observed. In comparison, CI listeners had BIFs that were less similar across listeners and across ears within the same listener. These results indicate that CI listeners use idiosyncratic listening strategies for speech perception that are ear-specific and cannot be fully accounted for by cochlear spread of excitation.

2aSC2. Intelligibility of British English for American younger and older adults with and without hearing loss. Caroline Champougny, Sarah H. Ferguson, and Sadie Schilaty (Commun. Sci. and Disord., Univ. of Utah, 390 S 1530 E, Rm. 1201 BEH SCI, Salt Lake City, UT 84112, u6002580@utah.edu)

In clinics, patients with hearing loss often report that they have difficulty understanding British-accented speech, despite having in the past watched and understood British television shows without trouble. Clopper and Bradlow (2008) examined the intelligibility of four United States regional dialects by American listeners, but to our knowledge, no study to date has quantified the intelligibility of non-U.S. varieties of English for American listeners. The present study will test the intelligibility of British English sentences by American younger and older adult listeners with normal hearing as well as a group of older adults with hearing loss. Participants will be presented with Basic English Lexicon (BEL) sentences produced by one American and one British talker; sentences will be presented in quiet and in a background of 12-talker babble. The resulting data will reveal how well hearing-impaired older adults understand British English, whether talker accent interacts with the presence or absence of noise, and whether older adults are disproportionately impaired in understanding British English compared to younger adults.

2aSC3. Perceptual speech intelligibility and speech production variability in Mandarin-speaking children with cerebral palsy. Li-mei Chen, Yu Ching Lin (Dept. of Foreign Lang., National Cheng Kung Univ., 1 University Rd., Tainan 701, Taiwan, leemay@mail.ncku.edu.tw), Katherine C. Hustad, and Raymond D. Kent (Waisman Ctr., Madison, WI)

Perceptual speech intelligibility index can provide an objective standard for evaluating how speech deficiencies affect listeners' judgments. This study investigated the effective variables for assessing speech intelligibility in Mandarin-speaking children with cerebral palsy. Acoustic measurements include vowel working space, vowel duration, pitch variation, and intensity variation of speech samples collected from picture naming tasks in two children with cerebral palsy (CP) and two typically developing children (TD) at four years old. Only clear word productions in terms of perceptual judgment and vowel formant display were incorporated to secure the reliability of data analysis. For speech intelligibility, eight judges were recruited to record the words they heard from the speech samples. Major findings are: (1) CP children had a smaller vowel space area due to limited control of tongue compared to TD children; (2) CP children showed longer vowel duration because they spent more time producing speech due to limited motor control; (3) CP children exhibited greater pitch and intensity variations due to instability in speech production. More data from more participants should be included for analysis to verify the preliminary findings.

2aSC4. Age-related differences in talker segregation and selection: Contributions of attention to voice features. William J. Bologna (Dept. of Hearing and Speech Sci., Univ. of Maryland, Medical University of South Carolina, 135 Rutledge Ave., MSC 550, Charleston, SC 29425, bologna@musc.edu), Jayne B. Ahlstrom, and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Focused attention on expected voice features, such as F0 and spectral envelope, may facilitate segregation and selection of a target talker in competing talker backgrounds. Age-related declines in attention may limit these abilities in older adults, resulting in poorer speech understanding in complex environments. To test this hypothesis, younger and older adults with normal hearing listened to sentences with a single competing talker. For a majority of trials, listener attention was directed to the target by a cue phrase that matched the target talker's F0 and spectral envelope; on these trials, younger adults outperformed older adults. On the remaining "probe" trials, the target's voice unexpectedly differed from the cue phrase in terms of F0 and spectral envelope; performance declined for both groups, and younger and older adults performed similarly. Thus, older adults performed poorer than younger adults only when attention could be focused on an expected voice. Moreover, older adults responded more frequently than younger adults with words from the competing sentence rather than the target sentence. Taken together, these results support the hypothesis that declines in the ability to focus attention on expected voice features contribute to speech understanding difficulties of older adults in complex environments. [Work supported by NIH/NIDCD.]

2aSC5. Speech produced in noise: Relationship between listening difficulty and spectral and durational parameters. Simone Graetzer, Pasquale Bottalico, and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, sgraetz@msu.edu)

Conversational speech produced in noise can be characterized by increases in intelligibility relative to conversational speech produced in quiet. The objectives of the study were to evaluate the listening difficulty of speech produced in different noise and style conditions, to evaluate the spectral and durational speech modifications that occurred in these conditions, and to determine whether the spectral or durational parameters predicted listening difficulty. Nineteen subjects were instructed to speak at normal or loud volumes in the presence of background noise at 40.5 dB(A) and babble noise at 61 dB(A). The speech signal was amplitude-normalised, combined with pink noise to obtain a signal-to-noise ratio of -6 dB, and presented to 20 raters who judged their listening difficulty. It was found that vowel duration, fundamental frequency (f_o, in semitones) and the proportion of the spectral energy in high relative to low frequencies increased with the level of the noise, independently of the effect of style. Listening difficulty was lowest when the speech was produced in the presence of high level noise and in the loud style, indicating improved intelligibility. The difference in spectral energy was observed to predict listening difficulty, and, therefore, intelligibility scores (*IS*, the percentage of words understood correctly).

2aSC6. Effect of auditory-motor mapping training and speech repetition training on consonant and vowel accuracy in minimally verbal children with autism spectrum disorder. Karen V. Chenausky, Andrea Norton, and Gottfried Schlaug (Neurology, Beth Israel-Deaconess Medical Ctr., Beth Israel-Deaconess Medical Ctr., 330 Brookline Ave., Boston, MA 02215, kvchenau@bidmc.harvard.edu)

Various therapies exist for teaching first words to minimally verbal (MV) children with autism spectrum disorder (ASD). Previous outcome measures have focused on number of words imitated or produced spontaneously, or on communication rate. No studies thus far have examined phonetic accuracy in MV ASD as a result of therapy, yet understanding whether therapy improves articulation in MV ASD is important for understanding how speech is affected in ASD and how best to treat these children. In this preliminary study of 30 children with MV ASD, we report on perceptual analysis of their speech after 25 sessions of one of two therapies, employing bisyllabic stimuli with a variety of consonant types. Twenty-three children received Auditory Motor Mapping Training (AMMT), where stimuli are intoned at approximately one syllable per second. Seven children, matched on age and cognition to 7 AMMT participants, received Speech Repetition Training (SRT), where non-intoned stimuli are presented at a normal speech rate. ANOVAs indicated that AMMT resulted in greater gains in consonant production accuracy than SRT. Both therapies produced equally significant improvement in vowel accuracy. Additional, ongoing analyses concern participants' accuracy for different consonant types, investigating whether accuracy is greater for earlier-appearing than later-appearing consonant types.

2aSC7. A longitudinal study of the effects of aging on speech breathing: Evidence of decreased expiratory volume in speech recordings. Simone Graetzer and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, sgraetz@msu.edu)

Age-related changes occur in speech that are associated with structural, physiological, and immunological processes involving the oral and nasal cavities, the larynx and pharynx, and the respiratory system. With aging, laryngeal tissues tend to degenerate or atrophy and laryngeal cartilages tend to ossify. These changes can lead to increased instability and perceived hoarseness or harshness, reduced loudness, and changes in fundamental frequency. In the respiratory system, a decline in lung and diaphragm elasticity and muscle strength can occur, and the thoracic cage can stiffen, leading to reductions in lung pressure and forced expiratory volume (hence, an increase in residual lung volume). In this study, recordings of three female and three male subjects were analyzed. These were made over the course of between 18 and 48 years (with a mean of 32.5 years). Samples of five minutes in length were extracted from each recording. Subsequently, trained raters measured the durations of exhalations during speech (termed "breath groups"). The results indicate decreases in breath group duration for most subjects as their age increased (especially from 65 years onwards), consistent with the decline in expiratory volume reported in the literature.

2aSC8. Evaluating how fine-grained changes in the spatial and temporal properties of audiovisual speech influence the perception of linguistic meter. Robert A. Fuhrman, Stanislaw Nowak, and Eric Vatikiotis-Bateson (Linguist, Univ. Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, robert.a.fuhrman@gmail.com)

The visual information that contributes to speech perception has been known for over a half century (Sumby and Pollack, JASA, 1954). More recent work has demonstrated two important constraints on the benefit of visual information to speech perception. First, the benefit patterns with temporal constraints on general audiovisual sensory processing, and degrades when auditory and visual signals are misaligned by an offset on the order of the duration of a syllable. Second, studies have demonstrated that low spatiotemporal frequency visual information is sufficient for boosting speech intelligibility. Collectively, these findings suggest that visual information facilitates processing of linguistic information organized in the speech stream primarily at the level of the syllable. Our experiments address how manipulation of the phasing and amplitude of visual components of the audiovisual stream associated with reiterant speech affects the perception of linguistic stress. Our analyses focus on determining the sensitivity of speech perception to changes in the fine-grained structure of visible motion (kinematics) and its alignment with the speech acoustics.

2aSC9. Vocal matching in interactions between mothers and their normal-hearing and hearing-impaired twins. Maria V. Kondaurova (Psychol. and Brain Sci., Univ. of Louisville, 699 Riley Hospital Dr. – RR044, Indianapolis, IN 46202, maria.kondaurova@louisville.edu), Laura C. Dilley (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), Tonya R. Bergeson-Dana (Otolaryngol. – Head & Neck Surgery, Indiana Univ. School of Medicine, Indianapolis, IN), and Mary K. Fagan (Commun. Sci. and Disord., Univ. of Missouri, Columbia, MO)

Vocal matching, the ability to imitate phonetic properties of speech, was examined in interactions between mothers and their normal-hearing (NH) and hearing-impaired twins who used hearing aids (HAs) or a cochlear implant (CI). Vocalizations of three mother-twin triads were recorded in three sessions over 12 months. In one triad, the twins were 15.8 months old and NH. In another triad, the twins were 11.8 months; one was NH while the other had HAs. In the third triad, both twins were 14.8 months; one was NH while the other had a CI. A vocal match was defined as an instance of perceptual and acoustic similarity between adjacent maternal and infant utterances in relation to pitch height and contour, utterance duration, rhythm, or vowels and consonants. Reciprocal vocal matching occurred in 28% to 38% of infant vocalizations across triads. At session three, CI and HA infants' reciprocal vocal matches increased compared to two previous sessions and to those of NH siblings; reciprocal vocal matches in the NH dyad decreased over time. The results suggest that vocal matching is a part of linguistic interactions between mothers and their NH and HI infants and that pediatric hearing loss affects mothers' and infants' imitative abilities.

2aSC10. Clear speech benefits for perception of center-only and edge-only syllables. Jenna S. Luque, Nathan Maxfield, Jennifer J. Lister, and Catherine L. Rogers (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, jennaluque@usf.edu)

Clear speech is a speaking style that has been shown to enhance intelligibility in noise; however, the underlying reasons for this enhancement are less well understood. One hypothesis is that listeners require less acoustic information to identify syllables spoken in clear than in conversational speech, allowing them to make use of briefer dips in fluctuating noise. The present study tests this hypothesis by presenting six /bVd/ syllables ("bead, bid, bed, bayed, bad," and "bod"), produced in clear and conversational speech styles, to 20 monolingual native English-speaking listeners in a six-alternative forced-choice task. These syllables were modified to present

varying portions of the syllable to listeners (20, 40, 60, or 80 ms of the syllable preserved). First, center-only syllables were created, in which acoustic information around the vowel midpoint was preserved. Second, edge-only syllables were created, in which information from the vowel center was silenced and formant transitions preserved, with vowel duration either maintained or equalized. Preliminary results show some clear-speech benefits in the partial-syllable conditions, with benefits mainly at shorter gates for center-only syllables and mainly at longer gates for edge-only syllables. Implications for understanding the source of clear-speech benefits in noise and phoneme perception more generally will be discussed.

2aSC11. The effect of talker and listener depressive symptoms on speech intelligibility. Hoyoung Yi (Commun. Sci. and Disord., The Univ. of Texas at Austin, 2504A Whitis Ave., Austin, TX 78712-0114, hoyoung@utexas.edu), Rajka Smiljanic (Linguist, The Univ. of Texas at Austin, Austin, TX), and Bharath Chandrasekaran (Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX)

It is widely recognized that depression is associated with deficits in communication behaviors (Miller, 1975), but few studies have examined speech intelligibility in talkers and listeners with elevated depressive symptoms. The present study examined intelligibility of conversational and clear speech sentences in the presence of speech shaped noise and one-talker babble noise. Talkers and listeners were young adults varying on the extent of depressive symptoms, and classified as having high depressive (HD) symptoms or low depressive (LD) symptoms based on a self-report scale (Radloff, 1977). The results showed that increased intelligibility through conversational-to-clear speech modifications was smaller for HD listeners than LD listeners during only one-talker babble noise which is consistent with Chandrasekaran *et al*. (2014). Intelligibility was scored lower for the sentences produced by HD speakers compared to LD speakers. Acoustic analyses indicate smaller conversation-to-clear speech modifications for speaking rate, energy in the 1-to-3 kHz, F0 mean, and F0 range for HD talkers compared to LD talkers. The results revealed that individuals with elevated depressive symptoms exhibit difficulty producing listener-oriented speaking style changes resulting in lower overall intelligibility. Adverse listening conditions involving informational masking, oft-present in typical social environs, present particularly challenging communication conditions for HD listeners.

2aSC12. The role of the carrier waveform in vocoded Mandarin speech perception with applications to cochlear implants. Jessica H. Schiltz (Speech & Hearing Sci., Arizona State Univ., 297 E LaSalle Ave., Apt. #: 205A, South Bend, IN 46617, jschiltz@nd.edu) and Visar Berisha (Speech & Hearing Sci., Arizona State Univ., Tempe, Arizona)

Cochlear implants (CI) enable patients to perceive sound by the generation and transmission of electrical impulses. Voice encoded (vocoded) audio is commonly used as a simulator of CI speech. Past studies of vocoded English concluded that coded speech intelligibility is independent of whether a sinusoidal or noise-based carrier is used. Based on this work, recent CI research for tonal languages like Mandarin, have opted for a sinusoidal carrier without considering the impact of the carrier on intelligibility. The importance of spectral cues in Mandarin speech necessitates further analysis of the relationship between the carrier and intelligibility. This study explored intelligibility differences between English and Mandarin vocoded speech. This approach assessed speech recognition of randomly presented phrases to normal hearing English and Mandarin listeners. Available frequency channels and carrier type were varied to compare their effects on Mandarin word and tone identification. Results indicated that audio processed with a sinusoidal carrier led to significantly lower Mandarin intelligibility scores. In comparison, no effect was observed between English and Mandarin intelligibility when a noise-based carrier was used. The data suggest that the nature of the carrier type affects tonal language intelligibility and warrants further research as an experimental consideration in vocoded speech studies.

2aSC13. Exploring the effects of masker spectro-temporal coherence on the informational masking of speech. Brian Roberts and Robert J. Summers (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, b.roberts@aston.ac.uk)

The effect of an extraneous formant on intelligibility is influenced by the depth of variation in its frequency contour. This study explored whether masker impact also depends on spectro-temporal coherence, using a method ensuring that interference occurs only through informational masking. Three-formant analogs of natural sentences were generated using a monotonous periodic source. Target formants were presented monaurally; the target ear was assigned randomly on each trial. A competitor for F2 (F2C) was presented contralaterally; listeners must reject F2C to optimize recognition. In the reference condition, F2C was created by inverting the F2 frequency contour and using a constant RMS-matched amplitude contour. In the coherent conditions, the F2C frequency contour was preserved but the amplitude contour was divided into abutting 100- or 200-ms segments using a raised-cosine envelope (10-ms rise/fall); these values were informed by typical syllable durations. Segment order was randomized in the incoherent conditions, introducing abrupt discontinuities into the F2C frequency contour. Adding F2C lowered keyword scores, but to the same extent for the reference, coherent, and incoherent conditions. This suggests that the impact on intelligibility depends critically on the overall extent of frequency variation in the interferer, but not on its spectro-temporal coherence. [Work supported by ESRC.]

2a TUE. AM

2aSC14. Spectral contrast effects in vowel categorization by listeners with sensorineural hearing loss. Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu) and Josh Alexander (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

The auditory system is highly sensitive to changes in acoustic input. This is especially true for sounds with relatively stable (reliable) spectral properties across time. Over a limited range, changes to the spectrum (e.g., spectral peak location upon introduction of a new sound) are perceptually enhanced in proportion to the property's long-term reliability, producing spectral contrast effects (SCEs). For example, a neutral vowel between /ɪ/-/ɛ/ is more likely to be labeled /ɛ/ (high-F_1) when preceded by sounds with a reliable low-frequency peak in the F_1 range for /ɪ/, and *vice versa*. Yet, it is unknown how SCEs affect speech perception by hearing-impaired (HI) listeners because research has only examined normal-hearing (NH) listeners. Here, listeners with mild-to-moderate HI identified target vowels varying from /ɪ/-/ɛ/ that followed a precursor sentence. Reliability of precursor spectral peaks was manipulated using low-F_1 or high-F_1 bandpass filters with +5 to +20 dB gain. SCE magnitude was proportional to precursor filter gain (like NH listeners), and surprisingly, to the amount of low-frequency hearing loss. Thus, mechanisms responsible for SCEs are not dependent on healthy hearing, and are magnified by auditory filter broadening associated with sensorineural hearing loss. Implications of these findings for speech perception will be discussed.

2aSC15. Intonation perception problems in advanced age. Amebu Seddoh, Afua Blay, and John Madden (Dept. of Commun. Sci. & Disord., Univ. of North Dakota, 290 Centennial Dr. Stop 8040, Grand Forks, ND 58202, seddoh@und.edu)

This study sought to determine whether intonation processing difficulties for elderly people are attributable to stimulus contextual factors. Participants were 20 old (61–84 years) and 15 young (19–29 years) adult native speakers of English. They were presented auditorily a total of 108 English sentences to decode following evaluation of their hearing abilities. Each sentence conveyed either positive or negative emotional meaning and was accompanied or unaccompanied by contextual information. Both groups of participants demonstrated significantly better outcomes on the decoding of the stimuli presented with contextual information (CS) compared to the decontextualized sentences (DCS). The elderly participants performed poorly relative to the young adults on the perception of positive and negative emotions in the DCS, as well as negative emotions in the CS. On the other hand, their performance on perception of positive emotion in the CS was comparable to that of the young adults. These findings are consistent with data that have shown that intonation meaning is context dependent

[e.g., Woodland and Voyer, Metaphor Symbol 26, 227–239 (2011)], and that elderly people have greater amygdala activation for positive than negative stimuli [Mather *et al.*, Psychol. Sci. **15**, 259–263 (2004)]. They suggest that intonation perception problems in old age may be influenced by stimulus contextual information, but part of their origin may lie in neurobiological factors such as age related changes in the brain.

2aSC16. Speech recognition thresholds in reverberation: Effects of age and hearing loss. Nirmal Kumar Srinivasan, Michelle R. Molis, and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, nirmal.srinivasan@va.gov)

While young normally hearing individuals can function in moderate amounts of reverberation with a minimal reduction in speech understanding, older individuals with and without hearing loss are much more susceptible to reverberant distortions of the speech signal (Harris and Reitz, 1985; Danhauer and Johnson, 1991; Helfer, 1992). In this experiment, we evaluated the individual contributions of age and hearing loss to speech reception thresholds (SRT) for reverberant speech (T60: 0, 300, 600, and 1000 ms) in the presence of speech-shaped noise. A one-up one-down adaptive-tracking procedure was used to measure SRT in the better ear of each of 27 listeners. Results indicated, as expected, that older-hearing-impaired listeners had higher SRTs as compared to younger and older normally hearing listeners. However, the variability among listeners was very high. Multiple regression analyses predicting SRT with age and hearing loss as factors indicated that hearing loss was the most significantly contributing factor for all four reverberation times tested. The effects of age and hearing loss on speech recognition in reverberation will be discussed in relation to the data available in the literature. [Work supported by VA RR&D I01RX001020.]

2aSC17. Reliability and time to administer visual analog scaling of intelligibility. Samantha T. Mocarski and Peter J. Watson (Speech-Language-Hearing Sci., Univ. of Minnesota - Twin Cities, 164 Pillsbury Dr., Shevlin 115, Minneapolis, MN 55455, pjwatson@umn.edu)

Measuring speech intelligibility is important in assessing the severity of impaired communication; and is used to demonstrate change overtime, e.g. before and after therapy. Orthographic transcription of speech is considered the clinical-gold standard of assessment, but can be time consuming. Many clinicians opt for quicker, subjective scaling such as Likert-interval, percent-estimate, and more recently visual-analog scaling (VAS). Percentage-estimate and Likert-interval scaling produce lower intra- and inter-judge reliability [Schiavetti (1992)] than transcription. Recently, VAS was used to study intelligibility in persons with dysarthria and found that reliability was good [Stipancic *et al*. (2015)] for mild deficits. We compared VAS to transcription under 4 SNR levels of background noise (−8, −5, −2, and 5). This approach allowed us to compare the reliability of VAS to transcription throughout a range of intelligibility levels. Twenty listeners both transcribed and visually scaled the intelligibility of audio-recorded IEEE sentences of a man and woman under the 4 SNR conditions. Reliability of VAS was as good as transcription at the extremes of intelligibility and less so in the middle levels. Transcription took 3x the time of VAS to administer.

2aSC18. Perceived emotion in clear speech: Effect of simulated hearing loss. Shae D. Morgan, Skyler G. Jennings, and Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1201, Salt Lake City, UT 84112, shae.morgan@utah.edu)

Previous research suggests that both young normal-hearing and older hearing-impaired listeners judge clear speech as sounding angry more often than conversational speech. Interestingly, older hearing-impaired listeners were less likely than young normal-hearing listeners to judge sentences as angry in both speaking styles, suggesting that age and/or hearing loss may play a role in judging talkers' emotions. An acoustic cue that helps distinguish angry speech from emotionally neutral speech is increased high-frequency energy, which may be attenuated or rendered inaudible by age-related hearing loss. The present study tests the hypothesis that simulating such a hearing loss will decrease the perception of anger by young normal-hearing listeners. Sentences spoken clearly and conversationally were processed and filtered to simulate the average hearing loss of the older hearing-impaired listeners from a previous study. Young normal-hearing listeners were asked to assign each sentence to one of six categories (happiness, sadness, anger, fear, disgust, and neutral). The judgments will be combined across listeners for each sentence, creating a percentage score for each emotion. The results from judgments of filtered stimuli will be compared with those of unfiltered stimuli as well as with results from young normal-hearing and older hearing-impaired listeners from previous studies.

2aSC19. "That sounds like me" Infants prefer vowels with infant vocal resonances. Linda Polka, Matthew Masapollo (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave., 8th Fl., SCSD, Montreal, QC H3A 1G1, Canada, linda.polka@mcgill.ca), and Lucie Menard (Dept. of Linguistics, Univ. of PQ at Montreal, Montreal, QC, Canada)

Recent research shows that infants listen preferentially to synthesized vowels that specify an infant source and resonance properties over vowels that simulate a adult female (Masapollo *et al*., 2015). Infants also preferred vowels with infant resonances over vowels with adult vocal resonances when f0 values were matched (210–240 Hz) across resonance types, suggesting that infant resonance is sufficient to elicit this preference. In this study, we investigate whether infants maintain a preference for infant vocal resonances when f0 values are modulated. In experiment 1, infants listened longer to vowels with infant formants and high f0 values (400–450 Hz) than vowels with adult formants and lower f0 (315–360 Hz). In experiment 2, the same preference emerged when f0 values were reversed; infants listened longer to vowels with infant formants and lower f0 values (315–360 Hz) than to vowels with adult formants and higher f0 (400–450 Hz). These findings show that infant resonance is sufficient to elicit this preference. Given that infants begin to produce vowel-like sounds at 3–4 months, these findings also support the "articulatory filter" hypothesis (Vihman, 1993) which claims that infants are perceptually biased toward speech that resembles their own vocal patterns.

2aSC20. Visual influences on the natural referent vowel bias. Matthew Masapollo, Linda Polka (McGill Univ., 1266 Pine Ave. West, Montreal, QC H3G 1A8, Canada, matthew.masapollo@mail.mcgill.ca), and Lucie Ménard (Univ. of PQ at Montreal, Montreal, QC, Canada)

Research indicates that perceivers (both adult and infant) are universally biased to attend to vowels with extreme articulatory/acoustic properties (peripheral in F1/F2 vowel space). Yet, the nature of this perceptual phenomenon (i.e., the natural referent vowel [NRV] bias) is not fully understood. The present research investigates whether this bias is attributable to general auditory processes or to phonetic processes that track articulatory information available across modalities. In experiment 1, we examined whether adult perceivers are biased to attend to visual information that specifies extreme vocalic articulations. As predicted by the phonetic account, we found a bias favoring relatively more peripheral vowels when only acoustic or only visual speech information was present. In experiment 2, we investigated how the integration of acoustic and visual speech cues influence the effects documented in experiment 1. When acoustic and visual cues were phonetically congruent, a peripheral vowel bias was observed. In contrast, when acoustic and visual cues were phonetically incongruent, this bias was disrupted. Collectively, these results are compatible with the view that the NRV bias is phonetic in nature—the speech processing system appears to be biased toward extreme vocalic gestures, which may be specified in the optic, as well as in the acoustic, signal.

2aSC21. Analysis of available and listener-received phonetic information in babble-masked consonant identification. Noah H. Silbert and Lina M. Zadeh (Commun. Sci. and Disord., Univ. of Cincinnati, French East 344, 3202 Eden Ave., Cincinnati, OH 45267, silbernh@ucmail.uc.edu)

Speech communication commonly occurs in the presence of multiple, non-target talkers. Previous work has shown that the amount of glimpsed target speech is a good predictor of overall intelligibility of babble-masked speech (Brungart *et al*., 2006, JASA) and that an automatic speech recognition system trained on glimpses closely approximates listener accuracy as a function of the number of babble talkers (Cooke, 2006, JASA). The present work uses a number of machine learning models to analyze the auditory information available in babble-masked speech and in available glimpses of babble-masked speech. Regularized Linear Discriminant, Support Vector Machine, and Naive Bayes classifiers were fit to modeled auditory representations of babble-masked CV syllables (with C = t, d, s, z, and V = a). The machine learning models outperformed human listeners substantially (>70% versus 54% accuracy, respectively). Analysis of predicted confusion patterns indicates that the Naive Bayes model most closely approximates human error patterns. The effects of variation in the local and absolute thresholds for glimpse calculation are explored with respect to overall accuracy and error pattern prediction in these machine learning models.

TUESDAY MORNING, 24 MAY 2016 SOLITUDE, 8:00 A.M. TO 11:10 A.M.

Session 2aSP

Signal Processing in Acoustics, Engineering Acoustics, and Underwater Acoustics: Acoustic Array Systems and Signal Processing I

Mingsian R. Bai, Cochair
Power Mechanical Engineering, National Tsing Hua University, No. 101, Section 2, Kuang-Fu Road, Hsinchu 30013, Taiwan

John R. Buck, Cochair
ECE, UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747

Chair's Introduction—8:00

Invited Papers

8:05

2aSP1. The effect of nearby scatterers on the gain achieved by an acoustic array. R. Lee Culver (ARL, Penn State Univ., PO Box 30, State College, PA 16804, rlc5@psu.edu)

The coherent processing of signals from multiple sensors in an array offers improvements in angular resolution and signal-to-noise ratio. When the array is steered in a particular direction, the signals arriving from that direction are added in phase, and any signals arriving from other directions are not. Array gain (AG) is a measure of how much the signal arriving from the steering direction is amplified relative to signals arriving from all other directions. This talk presents measurements along with supporting theory and simulation showing that as scatterer density increases and AG decreases, random phase shifts in individual sensor signals become larger and occur more often, and signal correlation among the sensor is reduced. The theory and numerical simulation linking scatterer density to the AG are in good agreement with the measurements up to the point where multiple scattering becomes important.

8:25

2aSP2. Differential beamforming with circular microphone arrays. Jacob Benesty (INRS-EMT, Univ. of PQ, Montreal, QC, Canada) and Jingdong Chen (Ctr. of Intelligent Acoust. and Immersive Communications, Northwestern PolyTech. Univ., 127 Youyi West Rd., Xi'an, Shaanxi 710072, China, jingdongchen@ieee.org)

Differential beamforming cannot only achieve high directional gain but also can form frequency-invariant beampatterns. Therefore, it has the great potential to be widely used in voice communications to process broadband speech signals. The performance of a differential beamformer in terms of directivity factor (DF), white noise gain (WNG), and frequency invariance of the beampattern depends on many factors including the array geometry, the number of sensors, the inter-sensor spacing, etc. In the literature, most studies on differential beamforming have been focused on linear microphone arrays and not much efforts have been devoted to other array geometries. This paper is devoted to the study of circular microphone arrays. In comparison with linear arrays whose spatial response is a function of the steering angle, circular microphone arrays can have the same spatial response along many different directions. The focal point of this paper is on differential beamforming with uniform circular microphone arrays. The main topics addressed include: (1) major properties of circular microphone arrays, (2) how to design differential beamformers with different beampatterns, (3) how to design differential beamformers with different orders, and (4) how to achieve a good control of the white noise amplification problem, high directional gains, and frequency-independent responses.

8:45

2aSP3. Estimating the spatial spectra of Gaussian processes with co-prime sensor arrays. John R. Buck and Kaushallya Adhikari (ECE, UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, johnbuck@ieee.org)

A co-prime sensor array (CSA) is a nonuniform line array formed by interleaving two undersampled uniform line arrays. The CSA requires fewer sensors to span the same aperture as a densely sampled uniform line array (ULA), allowing the CSA to match the resolution of the ULA for direction of arrival estimation of narrowband planewaves. However, each CSA subarray suffers from aliasing, or grating lobes, due to the spatial undersampling. Vaidyanathan and Pal (2011) proved that if the subarray undersampling factors are co-prime, the aliasing can be unambiguously resolved by multiplying the spatial spectra of the subarrays. This product spatial spectra is the spatial cross-spectral density between the arrays, and is an estimate of the spatial power spectral density (PSD). In this talk, we extend the classic results of Jenkins and Watts (1968) on the periodogram PSD estimator for Gaussian processes to obtain the product processor's bias for spatially wide-sense stationary processes, and the processor's covariance for spatially white Gaussians. Additionally, we demonstrate that the CSA's product PSD estimate is not necessarily positive definite. Consequently, the CSA product spectrum may fail to detect weak signals in the presence of strong interferers. [Work supported by ONR BRC Program.]

9:05

2aSP4. Time-space analysis, synthesis, and room response modeling of virtual-reality sound fields using microphone and loudspeaker arrays. Mingsian R. Bai and Yi Li (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan 30013, Taiwan, msbai63@gmail.com)

This paper proposes a unified framework to analyze and synthesize spatial sound fields. In the analysis phase, an unbaffled 24-element circular microphone array (CMA) is utilized to "encode" the sound field based on plane-wave decomposition, whereas in the synthesis phase, a 32-element rectangular loudspeaker array is employed to "decode" the above-encoded sound field based on pressure matching. Plane-wave decomposition can be implemented in two ways. First, a two-stage algorithm uses the minimum variance distortionless response (MVDR) beamformer to estimate the directions of significant plane-wave components. Next, the amplitude coefficients of plane waves are calculated using the Tikhonov regularization (TIKR) algorithm. Alternatively, a one-stage algorithm that treats the sound field as a great number of plane-wave components uniformly distributed at all angles can be used. In this case, either TIKR or convex (CVX) optimization can be utilized to solve the underdetermined problem for a minimum-norm solution or a minimum-cardinality solution. The same technique is also useful in establishing the response model for a reverberation room with all direct sounds and reflections coded into plane-wave components. In the synthesis phase, pressure matching is achieved by the inverse of the preceding room response model for rendering loudspeakers in the reproduction room.

Contributed Papers

9:25

2aSP5. Subarray compressive beamforming for midfrequency active sonar on a cylindrical array. Jonathan Botts (Appl. Res. in Acoust. LLC, 209 N. Commerce St., Ste. 300, Culpeper, VA 22701-2780, jonathan.botts@ariacoustics.com), Jason E. Summers, and Charles F. Gaumond (Appl. Res. in Acoust. LLC, Washington, DC)

Subarray compressive beamforming is adapted to a cylindrical array for midfrequency active sonar detection of low-Doppler targets in reverberant environments. Low Doppler targets are difficult to detect due to spreading caused by side-lobes and the width of the main lobe. Compressive beamforming and, in particular, subarray compressive beamforming will be shown to drastically mitigate Doppler spreading and interference from static objects. A straightforward application of compressive beamforming to simulated data results in nearly perfect suppression of spread due to sidelobes and good isolation of low Doppler targets from the zero Doppler ridge. Artificially subdividing the array allows for multiple fully resolved sparse estimates of the incident sound field. The set of estimates from the set of subarrays may be combined to form a single sparse estimate, often fully suppressing zero Doppler clutter. Performance of both compressive beamforming and subarray compressive beamforming will be demonstrated using simulated data having realistic Doppler spreading due to multipath and environmental sources of Doppler spreading. Performance will be compared to conventional beamforming methods on the cylindrical array. In addition to practical applications, some theoretical challenges with adaptation to the cylindrical array will be discussed. [Work supported by a NAVSEA Phase I SBIR award.]

9:40

2aSP6. Three-dimensional audio immersion: Development and human perception testing of a large scale, 128 loudspeaker surround sound system. Alexander Kern and Michael Roan (Mech. Eng., Virginia Polytechnic Inst. and State Univ., 111 Randolph Hall, Blacksburg, VA 24061, alkern@vt.edu)

Large-scale loudspeaker arrays for immersive audio are beginning to see increased interest in the research and entertainment industries. There are many algorithms to produce 3D sound fields. Although these algorithms are well developed, there is little published work that quantitatively compares their performance in terms of human perception for various numbers of speakers and speaker arrangements. In this talk, we will present listener perception test results using a 3D, 128 loudspeaker array. This loudspeaker array is part of the CUBE theatre at Virginia Tech. The CUBE is a 5-storey 50 ft. x 50 ft. black-box theatre that uses Dante^{TM} audio-over-ethernet to send signals to the 128 loudspeakers. Combined with a 3D motion tracking system and an immersive video vision system. The CUBE provides a very high-resolution test bed for human perception studies. Using speech intelligibility tests with stimulus phrases and various interferers and noise fields, listener tests will be performed for surround sound algorithms such as Higher Order Ambisonics, Vector Based Amplitude Panning, Wave Field Synthesis, and Higher Order HARPEX (High Angular Resolution Plane Wave Expansion). The Higher Order HARPEX decoder is a novel improvement to First Order HARPEX. Also, we will present technical details of the development of a 16-element soundfield microphone.

9:55–10:10 Break

10:10

2aSP7. Partially correlated source decomposition techniques from a beamforming analysis. Blaine M. Harker, Kent L. Gee, and Tracianne B. Neilsen (Dept. Phys. & Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@gmail.com)

Recent improvements have extended the capabilities of beamforming methods from the uncorrelated monopole assumption to enable reconstructions of extended, partially correlated sources. Resulting source information contains both level and coherence properties that can predict the radiated sound field. In addition, the source cross-spectral matrix, while accurate, can in many cases be reduced into a more convenient or physically meaningful set of basis vectors. For example, a wavepacket is a self-coherent distribution that may be used to create a minimal set of mutually orthogonal basis vectors representing the source levels and coherence properties. In this study, a complex, extended, and partially correlated source is modeled, and the simulated near-field measurements are analyzed with inverse techniques (i.e., beamforming). Possible decompositions are evaluated for their ability to accurately represent the source properties, as well as reliably predict the field levels and coherence properties using the fewest basis terms. Decompositions examined include proper-orthogonal decomposition and spatial decomposition based on coherence lengths. The results point towards an improved interpretation of the source reconstructions and methods for predicting the noise environment from an extended, partially correlated noise source. [Work supported by ONR and USAFRL through ORISE.]

10:25

2aSP8. Azimuth-elevation direction finding using one four-component acoustic vector-sensor spread spatially as a parallelogram array. Yang Song (Universitat Paderborn, Paderborn, North Rhine - Westphalia, Germany) and Kainam T. Wong (Dept. of Electron. & Information Eng., Hong Kong Polytechnic Univ., DE 605, Hung Hom KLN, Hong Kong, ktwong@ieee.org)

An acoustic vector-sensor (a.k.a. a vector hydrophone) consists of three uni-axial velocity-sensors (which are oriented perpendicularly with regard to each other) and one pressure-sensor. Song and Wong have demonstrated how to space these four component-sensors apart in three-dimensional space, in order to extend the overall spatial aperture spanned by them, while improving the accuracy in estimating the azimuth-elevation direction-of-arrival of an acoustic emitter incident from the far field. This paper will focus on a special spatial geometry—where the four component-sensors occupy the four corners of a parallelogram in three-dimensional space.

10:40

2aSP9. Complementary arrays above and below the ocean surface. Paul Hursky (HLS Res. Inc., 12625 High Bluff Dr., Ste. 211, San Diego, CA 92130, paul.hursky@hlsresearch.com)

An experiment was performed off the coast of San Diego to study propagation from an aircraft at altitude to a microphone array above the ocean surface and to hydrophone arrays below the ocean surface. An acoustic hailing device was mounted on a small airplane. An in-air array was formed by mounting individual microphones to the frame of a small boat. Several hydrophone arrays were deployed from the boat. The airplane flew lawn mower patterns over these arrays transmitting broadband waveforms at several altitudes. We will discuss propagation modeling, array calibration and beamforming for this configuration. Doppler is a significant factor due to the high speed of the aircraft and due to the speed of sound in air being five times less than in water. We will assess matched filter gain on the air and air-water paths and discuss phenomena that degrade such gains.

10:55

2aSP10. Improving ray-based blind deconvolution of random shipping sources with short arrays in an ocean waveguide using adaptive beamforming. Juan Yang (Inst. of Acoust.,Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, yang_juanacoustics@163.com), Sung-Hoon Byun (Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea), and Karim G. Sabra (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

This paper introduces an improved ray-based blind deconvolution(RBD) algorithm for sources of opportunity, such as shipping noise, recorded on a short vertical line array (VLA) by using adaptive beamforming. The original RBD algorithm [Sabra *et al*., JASA, 2010, EL42-7] relies on first estimating the unknown phase of the random source using broadband beamforming along a well-resolved ray path to estimate the full channel impulse responses (CIR) between the unknown source and the VLA elements (up to an arbitrary time-shift) as well as recovering the radiated signal by the random source. Hence, RBD performance is limited by the VLA ability to separate adjacent ray path, especially for the case of small aperture. To address this limitation, broadband MVDR is used as a high resolution estimator to improve the RBD performance when using single snapshot recordings and short VLAs. To do so, smoothing techniques are used to increase the rank of cross-spectral data matrix since the ray arrivals emanating from the same random source are naturally correlated in an ocean waveguide when using single snapshot recordings. The improvement of this proposed RBD algorithm will be demonstrated using both numerical simulation and at-sea shipping noise data recorded in shallow water.

Session 2aUW

Underwater Acoustics: Target Physics and Scattering

Brian T. Hefner, Chair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Chair's Introduction—8:30

Contributed Papers

8:35

2aUW1. Target-depth estimation in active sonar. Alexis F. Mours (Gipsa-Lab, Université Grenoble Alpes, 11 rue des Mathématiques - BP 46, Saint Martin d'Hères Cedex 38402, France), Nicolas Josso (Thales Underwater Systems, Valbonne, France), Jérôme I. Mars, Cornel Ioana (Gipsa-Lab, Université Grenoble Alpes, Saint Martin d'Hères Cedex 38402, France, jerome.mars@gipsa-lab.grenoble-inp.fr), and Yves Doisy (Thales Underwater Systems, Valbonne, France)

This paper presents a new target-depth estimation method that is based on ray back-propagation with a probabilistic approach. This localization algorithm tries to minimize the mean-squared error of elevation angles at the receiver and arrival times between a model and measures. This method was tested on Monte-Carlo simulations of classic active sonar scenarios and using experimental data from a real tank. In active sonar with a point-target model, combined acoustic paths may exist. These have a different path between the sonar-target and the target-array. This paper discusses also about this ray identification. Simulations with vertical array suggest that the target-depth estimation can be realized with a low uncertainty compared to the water column for long ranges in a Mediterranean sound speed profile. However, some environmental parameters as random sound-speed profile, array depth or array tilts could increase the bias and the variance of the target-depth estimator. Results on experimental data with surface noise reveal a good estimation of the target depth and validate our localization algorithm on a constant sound-speed profile with a vertical array.

8:50

2aUW2. Three-dimensional underwater target localization with circular vector sensor array in strong reverberation environment. Ge Yu and Shengchun Piao (College of Underwater Acoust. Eng., Harbin Eng. Univ., Bldg. 145, Nantong St., Nangang District, Harbin 150001, China, liz.221@163.com)

As reverberation can be regarded as an output of time-varying stochastic filtering of the emitted signal, it always leads to high false alarm in shallow water. In this case, some methods have been used to suppress reverberation such as empirical mode decomposition filter, match filter, etc. However, a strong correlation between transmitted signal and reverberation restricts the performance of these methods. In this paper,a novel method based on a circular vector sensor array is proposed to detect and locate multiple targets, respectively, in a strong reverberation environment by using the difference of the Doppler shift distribution between reverberation and target echo. First, spherical harmonic decomposition and beamforming are used to process the wideband signal and generate multiple beams which contain spatial information of interested region. Then, fractional Fourier transform is adopted to get Doppler shift distribution and localization information of reverberation and target echo. Multiple targets can be detected and located separately by estimating variance fluctuation associated with the reflections in different beams. Simulation result shows that this method performs efficiently and reliably in server reverberation environment without any prior information.

9:05

2aUW3. Target strength observations of wobbly bubbles. Alexandra M. Padilla (School of Marine Sci. and Ocean Eng., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, apadilla@ccom.unh.edu), Kevin Rychert, and Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

Large methane bubbles released from shallow water seabed are of interest because bubbles facilitate the transport process of gas through their journey in the water column, making them more susceptible to reaching the atmosphere than those released from deep water. Several models that relate bubble radius and acoustic backscattering for ideal, spherical bubbles have been developed and are typically used by researchers attempting to invert acoustic backscatter measurements for bubble size and/or gas flux. However, based on theory and field experimentation, it has been shown that large free gas bubbles in liquids (large Eötvös and/or large Reynolds numbers) are non-spherical. A shallow water (<6 m) tank experiment has been designed to understand the influence of bubble shape on acoustic backscattering. This experiment was conducted for bubbles between 1 and 5 mm in radius (Eötvös number between 0.5 and 13, Reynolds numbers between 300 and 1500, and a Morton number of approximately $7.2 \times 10-^{11}$) and at frequencies between 10 and 100 kHz. These frequencies are well above bubble resonance but cover the range of typical echo sounder measurements of bubbles in the field. Calibrated bubble target strength for these bubbles will be compared to existing acoustic backscattering models for bubbles.

9:20

2aUW4. Theoretical and experimental study of a thin elastic cylindrical shell subjected to a point source. Soheil Shah-Hosseini, Fernand Leon, Farid Chati, Dominique Decultot, and Gerard Maze (Laboratoire Ondes et Milieux complexes, UMR CNRS 6294 - Université du Havre, 75, rue Bellot, Le Havre, Haute-Normandie 76600, France, soheil.shahhosseini@gmail.com)

The aim of this work is to study theoretically and experimentally, the acoustic radiation of a cylindrical elastic shell excited by a point source with no internal loading and immersed in a water tank. We are primarily focused on the theoretical study of a cylindrical elastic shell receiving a point force in the form of a Dirac pulse and the acoustic pressure radiated along the tube into a point at far-field observation. The model used is based on the theory of elasticity. It is therefore interesting, in a second step, to conduct experiments to validate these theoretical calculations and to realize complementary measures corresponding to the case of a focalized observer. The used shell is a cylindrical stainless steel tube characterized by its ratio equal to 0.94. Two experiments are carried out. The first experiment is based on a bistatic method, and the transmission and reception are performed by two focalized transducers, characterized by a center frequency of 1 MHz. For the second experiment, the two transducers are different in nature; the transducer acting as the receiver is replaced by a plane transducer characterized by a center frequency identical to the transmitter focalized transducer

9:35

2aUW5. Modes of targets in water and on sand driven by modulated radiation pressure of focused ultrasound. Timothy D. Daniel, Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu), Ivars Kirsteins (NUWC, Newport, RI), and Ahmad T. Abawi (HLS Res., La Jolla, CA)

In a previous work, we had shown that low frequency flexural modes of a circular plate could be excited using modulated radiation pressure generated by focused ultrasound [J. Acoust. Soc. Am. **137**, 2439 (2015)]. Recently, we conducted another experiment where we examined plate, spherical, and cylindrical targets with a larger, higher power focused ultrasound, transducer operated at a carrier of 0.5 or 1.5 MHz that was capable of producing much stronger acoustic radiation pressures. A laser vibrometer was acquired to obtain quantitative measurements of the surface velocity of the targets to facilitate analysis of the radiation force generated on the targets as well as the pressure levels of the target sound emissions. The vibrometer was able to identity modes as well as measure mode shape. Furthermore, we were able to estimate the amount of radiation force that was driving the targets by using the vibrometer measurements to calculate their radiation loading from the water. We were also able to measure the response of targets proud on sand, compared to previous "free-field" experiments, and detect resonant modes and scan the source to infer mode shapes of proud targets with both a hydrophone and vibrometer. [Work supported by ONR.]

9:50

2aUW6. Circular synthetic aperture sonar images of backscattering enhancements on solid elastic cubes due to Rayleigh waves and other mechanisms. Viktor Bollen (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, viktor.bollen@wsu.edu), Daniel S. Plotnick (Acoust., Appl. Res. Lab., Seattle, WA), and Philip L. Marston (Washington State Univ., Pullman, WA)

Aspect dependent backscattering enhancement mechanisms aid in detecting and identifying underwater objects. This work examines enhancement mechanisms on a solid metal cube observed in acoustic images, emphasizing Rayleigh type surface waves that undergo retroreflection at the corners of the face of the cube [K. Gipson and P. L. Marston, J. Acoust. Soc. Am. **105**, 700–710 (1998)]. Backscattering from a steel cube insonified in water was measured using a circular synthetic aperture sonar system. Image reconstruction was performed using Fourier based algorithms. Rayleigh waves are launched on a face of the cube when the local incident angle is equal to the Rayleigh coupling angle. The presence of four right-angle retroreflectors on each face means that multiple reverberations of this wave are observed. In reconstructed images of the cube, the Rayleigh wave mechanism is responsible for some of the brightest features. If the relative orientation of the circular aperture and the cube is such that specular glints are suppressed, edge and corner diffraction features are clearly observed. This relative orientation may be deduced from the reconstructed image. Discussions of both the spectral response and temporal response of the cube as a function of viewing angle will also be included. [Work supported by ONR.]

10:05–10:20 Break

10:20

2aUW7. Frequency response of proud and partially exposed spheres at flat interfaces. Aaron M. Gunderson and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, aaron.gunderson01@gmail.com)

The backscattering spectrum of spheres at flat interfaces viewed at small grazing angles depends upon reflections from the sphere's surface and the interface, as well as the sphere's elastic response and Franz wave contributions. For partially exposed spheres, one approach to modeling the reflective contribution is to extend the Kirchhoff approximation [K. Baik and P. L. Marston, IEEE J. Ocean. Eng. **33**, 386–396 (2008)] to spheres. Laboratory measurements were carried out using short tone bursts for solid spheres at air-water and sand-water interfaces in such a way that it was typically possible to distinguish between the reflective and delayed elastic contributions viewed in the time domain. For spheres at an air-water interface, the reflective contributions have strong similarities with the Kirchhoff approximation while the strength of the delayed elastic contributions depends strongly on the extent of exposure of elastic guided wave coupling regions. The spectra of proud targets have similarities with image approximations, yielding ridges and valleys in the frequency versus grazing angle domain (associated with interference of contributions of similar magnitude), modulated by elastic contributions. Elastic contributions from small spheres near an air-water interface display a Lloyd's mirror pattern. [Work supported by ONR.]

10:35

2aUW8. Effects of vertical obliquity on sonar images and elastic mechanisms for cylinders near a flat interface. Daniel Plotnick (APL, Univ. of Washington, 1510 NW Turner Dr., Apt. 4, Pullman, WA 99163, dsplotnick@gmail.com) and Philip Marston (Washington State Univ., Pullman, WA)

The backscattered signal from targets located near a flat interface is highly dependent on the relative positions and orientations of the acoustic source/receiver, the interface, and the target. The acoustic backscattering spectrum versus aspect angle, also called the "acoustic color'' or "acoustic template,'' of solid cylinders was previously studied for the case where the cylinder axis was vertically oblique relative to a nearby water-air interface [D. Plotnick *et al*., J. Acoust. Soc. Am. **137**, 470–480 (2015)]. The presence of the interface allows for multiple orientation-dependent paths by which sound is backscattered; this work examines the effects of vertical obliquity on these paths. Experimental data were on a solid cylinder at various obliquities near an air-water interface collected using a circular synthetic aperture sonar system. Emphasis will be placed on the effects of vertical obliquity on features in reconstructed sonar images. Several robust orientation dependent features are considered and the physical mechanisms identified through geometric arguments. Information about a target's 3-D orientation and size may be gleaned from these images, aiding the interpretation of features within the target's acoustic template. The coupling conditions for surface elastic waves are also considered. [Work supported by ONR.]

10:50

2aUW9. A comparison of the predictions of two competing sediment acoustic models for the scattering from inclusions in a sandy sediment. Anthony L. Bonomo and Marcia Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

Currently, there are several competing models that have been used to describe the acoustic properties of sandy sediments. These models include those that assume the sediment to behave as an acoustic fluid, a viscoelastic solid, and as a porous medium following Biot theory. Perhaps the two most sophisticated acoustic models that have been applied to sand are the Viscous Grain Shearing (VGS) model of Buckingham and the Extended Biot (EB) model of Chotiros. While both of these models have been used to fit measured sound speed dispersion and attenuation data, the reflection and scattering predictions made using these models differ. In this work, the finite element method is used to compare the scattered pressure predictions of these two models for the case of a sand half-space with a single inclusion. The geometry is assumed axisymmetric, and the inclusions are allowed to be either a fluid or an elastic solid. [Work supported by ONR, Ocean Acoustics.]

11:05

2aUW10. Modeling of sub-bottom volume scattering from turbidite sequences. Derek R. Olson and Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, dro131@psu.edu)

Abyssal sediments near the continental margin are often formed through turbidite deposition, which creates alternating layers of silt and mud. Modeling of reflection and sub-bottom volume scattering from turbidite sequences has been previously studied using an effective medium to model the propagation within the sediment, which is only applicable when the acoustic wavelength is much larger than the average layer thickness in the seafloor, and thus is limited to low frequencies. A volume scattering model is

developed here to extend the frequency range of previous models of scattering from turbidite sequences. This model employs the Born approximation, takes into account the correct propagation physics in a layered environment, and uses a point source as an incident field. Model results will be compared to previously reported scattering data collected in the Ionian sea.

11:20

2aUW11. Statistics of near surface underwater scattered sound including wave induced Doppler. Sean Walstead (Sensors and Sonar Systems, Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, sean.walstead@navy.mil)

The interaction of underwater sound with the ocean surface is explored. Analytic expressions for the Doppler shift and Doppler spread of surface reflected signals are developed. Statistical models for surface scattered intensity, arrival time, and Doppler fluctuations are developed in relation to wave shape, transducer position, and transmission frequency. Measurements from two experiments are considered. Data collected from AUTEC explore surface scattering effects in the frequency regime of 8 kHz–10 kHz, while data collected in a variable speed wind wave channel at Scripps Institution of Oceanography explore scattering statistics in the 200 kHz–400 kHz (VHF) frequency regime. Physics based analytic expressions for the equations of motion of specular reflection points (SRPs) along a time-evolving wave surface are derived. The time-varying Doppler shift of VHF underwater acoustic signals is attributed to SRP motion associated with surface wave propagation. Model predicted values are compared to experimental measurements and differences between the two tested frequency regimes are highlighted. This work has direct application to the improved performance of phase coherent underwater acoustic communications systems and the remote sensing of gravity-capillary waves.

Meeting of the Standards Committee Plenary Group

to be held jointly with the meetings of the

ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for:
ISO/TC 43, Acoustics,
ISO/TC 43/SC 1, Noise,
ISO/TC 43/SC 3, Underwater acoustics
ISO/TC 108, Mechanical vibration, shock and condition monitoring,
ISO/TC 108/SC 2, Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures,
ISO/TC 108/SC 3, Use and calibration of vibration and shock measuring instruments,
ISO/TC 108/SC 4, Human exposure to mechanical vibration and shock,
ISO/TC 108/SC 5, Condition monitoring and diagnostics of machine systems, and
IEC/TC 29, Electroacoustics

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

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

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

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

D. J. Evans, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 3 Use and calibration of vibration and shock measuring devices
13 Watch Hill Place, Gaithersburg, MD 20878

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

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

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

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

The meeting of the Standards Committee Plenary Group will follow the meeting of Accredited Standards Committee S2, which will be held on Monday, 23 May 2016 from 5:00 p.m. - 6:00 p.m.

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

Tuesday, 24 May 2015	11:00 a.m. - 12:15 p.m.	S12, Noise
Tuesday, 24 May 2016	2:00 p.m. - 3:00 p.m.	ASC S3/SC 1, Animal Bioacoustics
Tuesday, 24 May 2016	3:15 p.m. - 4:30 p.m.	ASC S3, Bioacoustics
Tuesday, 24 May 2016	4:45 p.m. - 5:45 p.m.	ASC S1, Acoustics

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

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

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

Meeting of Accredited Standards Committee (ASC) S12 Noise

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

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

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

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1, Noise, and ISO/TC 43/SC 3, Underwater acoustics, take note that the meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 24 May 2016..

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

TUESDAY AFTERNOON, 24 MAY 2016 SNOWBIRD/BRIGHTON, 1:00 P.M. TO 5:50 P.M.

Session 2pAA

Architectural Acoustics, Noise, and Engineering Acoustics: Predicting and Controlling Heating, Ventilating, and Air Conditioning Systems Noise

Robert C. Coffeen, Cochair
Architecture, University of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045

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

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. Heating, ventilating, and air conditioning systems noise and electronic sound masking systems in buildings. Andrew N. Miller (BAi, LLC, 4006 Speedway, Austin, TX 78758, amiller@baiaustin.com)

HVAC systems are not typically adequate for both sound masking and neutral background noise quality. Electronic sound masking system cost and overall background noise quality can benefit from considering HVAC system noise character when designing sound masking systems. The contributions of adding electronic sound masking to neutralize background noise character generated by HVAC systems in buildings are explored. Results from general investigations on the topic are presented. General relationships of background noise to speech and noise reduction of construction assemblies are discussed.

1:25

2pAA2. Heating, ventilation, and air conditioning noise calculations after the slide rule and calculator. Steven Thorburn (Thorburn Assoc. Inc., 20880 Baker St., PO Box 20399, Castro Valley, CA 94546, sjt@TA-Inc.com)

Computer programs that predict HVAC system noise are now common. But, what was the basis of the design of the programs, what are the noise control assumptions? This paper will present how the 1984 ASHRAE Chapter 37 Sound and Vibration from the Systems Handbook was converted into a computer program and the limitations that were found in the chapter and how they were modified to more accurately predict the HVAC system noise and it control. This program has been updated 4 times over the years and is now an EXCEL bolt on.

1:45

2pAA3. Sound data for a new format of short-profile fan coil units. K. Anthony Hoover and Brandon W. Cudequest (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

In recent years, a new format of short-profile fan coil units as offered by several manufacturers has become increasingly popular. The sound data for such units are generally plotted in octave bands on NC curves, and have the appearance of great data. However, any scrutiny raises a variety of concerns. Fundamental methods were used to approximate power levels from the provided pressure levels for use in HVAC system noise calculations on various projects. In July 2015, test lab power level data were serendipitously provided for two fan coil units that had been included in the design for a new project. In due course, test lab data for six units, based on AHRI Standard 260, were obtained. This paper will discuss the approximation methods, ranges of differences between pressure level and power level data, effects on results of calculations, referenced measurement standards, and an issue related to pure tones.

2:05

2pAA4. A comparison of three manufacturer's mechanical noise prediction programs and how estimated noise levels from each program compare to field measurements in the subject rooms. Eric McGowan and Robert Coffeen (The Univ. of Kansas, 1465 Jayhawk Blvd., Lawrence, KS 66045, eric.mcgowan789@gmail.com)

The estimation of noise due to heating, ventilation, and air conditioning systems using a program supplied by a manufacturer is a common practice in acoustical consulting. There is little published research comparing these programs against each other and comparing the noise levels predicted by these programs against actual measurements taken in the field. Three rooms on campus at The University of Kansas and one multipurpose room off campus were modeled as similarly as possible in these programs using as-built mechanical drawings and manufacturer's noise data. In addition, ambient noise measurements were taken in each space with the mechanical system operating at a "normal" capacity for springtime in Kansas. The results were used to attempt to determine how the programs compared to each other such as which one was most conservative and which program consistently came closest to predicting the actual noise levels measured in the field.

2:25

2pAA5. Status and dissemination of the current calculation algorithms used in the American Society of Heating and Refrigeration Engineers handbooks. Gregory A. Miller (Threshold Acoust., LLC, 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, gmiller@thresholdacoustics.com)

The Fundamentals and Applications Handbooks by the American Society of Heating and Refrigeration Engineers (ASHRAE) include chapters on noise and vibration control that provide the foundation of most HVAC noise control calculations, whether in commercially available software or in custom software developed by practitioners. The handbooks, however, do not always delve into the detailed calculation algorithms that were used to develop the published tables and charts nor do they provide detailed descriptions of the assumptions and limitations of various calculation methods. ASHRAE last published a compendium of all the calculation algorithms in 1991; the handbooks have been updated to reflect recent research, but without a corresponding update of the detailed algorithms. This paper will summarize ways in which the algorithms have changed since 1991 and discuss the nascent work by the ASHRAE Technical Committee on Sound and Vibration Control (TC 2.6) to compile a new updated compendium of algorithms.

2:45

2pAA6. How audible tones affect psychoacoustic perception of heating, ventilation, and air conditioning noise. Joonhee Lee and Lily M. Wang (Durham School of Architectural Engineering, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, joonhee.lee@huskers.unl.edu)

Assessment of noise from heating, ventilation, and air-conditioning (HVAC) equipment using current building noise rating systems like Noise Criteria or Room Criteria is not reliable when the noise signals include audible tones. Equipment manufacturers and the acoustical consulting community have therefore been seeking new tone criteria for HVAC noise. This paper reviews the current knowledge base, regarding perception of tones and development of tone criteria. Two recent subjective investigations by the authors are presented. The first study utilized multidimensional scaling analysis (MDS) to identify key psychoacoustic parameters to predict human responses to HVAC noise. The second study aimed to develop a more precise method to quantify annoyance perception when multiple tones are present. Results have been used to develop an annoyance prediction model for HVAC noise with tones, which may be used to set evidence-based tone criteria.

3:05

2pAA7. Predicting airborne sound levels from mechanical equipment rooms. Felicia Doggett (Metropolitan Acoust., LLC, 40 W. Evergreen Ave., Ste. 108, Philadelphia, PA 19118, f.doggett@metro-acoustics.com)

Predicting airborne sound levels from mechanical equipment rooms or rooftop equipment to adjacent occupied spaces has traditionally been a bit of a gray area when it comes to modeling. Over the last two decades, we have attempted to refine our in-house tools to predict sound levels accurately, but there is much information from various sources and it has been difficult to know what is right and what is not so right. The equation NR = TL + 10log(a/S) only applies to reverberant rooms; other equations from Long, Rathe, among others attempt to refine this equation for other conditions including separating out direct and reverberant sound fields. This presentation explores these various modeling techniques and raises the question as to which one may be "most correct" to use for various conditions and situations.

3:25

2pAA8. Linking design decisions to speech intelligibility. Andrew Hulva, Michael Ermann, Kirsten Hull, Randall Rehfuss, Aaron Kanapesky, and Alex Reardon (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu)

What is the largest classroom that will support unamplified communication? What design attributes and acoustical conditions contribute to speech intelligibility, and how much should each design attribute be weighted when predicting speech transmission index (STI)? This line of inquiry aims to correlate room design and mechanical equipment (HVAC) design to speech intelligibility (STI) in university lecture rooms. It seeks to link the STI of classrooms at Virginia Tech to ambient noise levels, reverberation time, impulse response characteristics, room volume, number of seats, distance to nearest HVAC terminal device, receiver location, and speaker source power. Ambient noise measurements and STI will be mapped on a 1 m x 1 m grid for a subset of the measured rooms to visualize the noise on the listener plane and better estimate the haptic component of signal-to-noise ratio in the spaces. The research also will explore quirks in the STI measurement itself, including differences between the direct and indirect measurement methods, differences permitted for test speech level, and differences from seat to seat within a single room.

3:45–4:00 Break

4:00

2pAA9. Using mechanical noise prediction software to provide suitable solutions. Kevin Butler (Henderson Engineers, Inc., 8345 Lenexa Dr., #300, Lenexa, KS 66214, kevin.butler@hei-eng.com)

Mechanical noise prediction softwares have become a useful tool for consultants to provide noise control recommendations. Two recent projects involving duct breakout will be analyzed to discuss how mechanical noise prediction softwares were used to analyze multiple solutions in order to provide a suitable solution for a medical office building and an open office space.

4:20

2pAA10. Noise reduction of heating, ventilating, and air conditioning systems in previously constructed buildings, primarily focusing on health facilities. Logan D. Pippitt (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, ldpippitt@gmail.com)

Excessive heating, ventilating, and air conditioning system noise is a leading cause of much unwanted noise throughout most commercial buildings, some of which are medical facilities with multiple systems implemented to meet the heating, cooling, and ventilation requirements. Even so, multiple opportunities for correction can be found. Many new buildings are being properly constructed with this problem in mind, though this still leaves many issues in previously constructed buildings and facilities where the noise issues have not been addressed. So what can be done? This paper will explore the possible heating, ventilating, air conditioning, and room acoustic noise issues of a particular nursing home in Olathe, Kansas. The building administrator receives complaints from families, residents, and staff members on multiple noise issues that can be directly attributed to HVAC system noise. Issues such as, but not limited to, exterior noise pollution, supply and return HVAC noise, duct breakout noise, fan coil unit noise, roof top unit source noise, and related room acoustics. In addressing these issues, solutions will aim to meet current ASHRAE acoustical guidelines for medical facilities. [This research was conducted through the School of Architecture, Design and Planning at the University of Kansas.]

4:40

2pAA11. Predicting and controlling heating, ventilating, and air conditioning systems noise: A comparison of computer noise control prediction tools and their potential pitfalls. Nicole Cuff (Acentech, 33 Moulton St., Cambridge, MA 02138, ncuff@acentech.com)

Appropriate background noise levels for mechanical systems are critical for a successful use of any project. There are several computer software packages that acousticians use to evaluate the background noise levels in a space; of course, the acoustician needs to understand all of the inputs into the programs to fully understand the outputs. We will discuss strategies to use these noise prediction tools successfully, including how to evaluate noise sources and translate manufacturer data to input into the programs, how to evaluate various noise paths together for one space, and how to consider sound propagation of different noise sources. We will compare several different and widely available computer noise prediction programs and discuss the positives and challenges of using each one, including the author's assessment of which circumstances lend themselves best to one program versus another.

5:00

2pAA12. Insertion loss provided by several commercially available air handling duct wrap materials that are applied to reduce sheet metal duct breakout noise levels. Robert C. Coffeen (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, coffeen@ku.edu)

Noise breaking out of sheet metal HVAC duct is often a significant and disturbing noise source for a building space being served by a supply or return duct or through which the supply or return duct is passing. Reducing breakout noise is often attempted by using heavy loaded vinyl duct wrap applied directly to the duct or applied over glass fiber insulation or foam sheet. Duct wrap insertion loss data were collected using a portion of a sheet metal duct with noise produced by a loudspeaker assembly suspended within the duct. Breakout noise levels were measured on adjacent sides of the duct with and without duct wrap. This approximate insertion loss data were obtained by a University of Kansas architecture student as a portion of work leading to a Master of Arts in Architecture degree.

Contributed Papers

5:20

2pAA13. Simulation of noise propagation through heating ventilation and air conditioning ductwork. David W. Herrin and Kangping Ruan (Dept. of Mech. Eng., Univ. of Kentucky, 151 Ralph G. Anderson Bldg., Lexington, KY 40506-0503, dherrin@engr.uky.edu)

Noise primarily propagates through the airspace or breaks out through the walls of HVAC ductwork. Finite element analysis was used to assess the noise attenuation of each of these paths. Specifically, the insertion loss and breakout transmission loss were assessed for both bare and lined ductwork. Sound absorptive lining was modeled using poroelastic finite elements and the ductwork itself was modeled using shell and beam elements. A diffuse field was approximated at the source using a collection of monopole sources having random phase. Results were compared to measurement with good agreement. Several conclusions can be made regarding noise transmission in large ducts that should be transferable to other industries.

5:35

2pAA14. Noise analysis of heating, ventilating, and air-conditioning systems: Modeling in the Trane acoustics program. Jennifer E. Russell (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, jrussell@siebeinacoustic.com)

Many different acoustical issues can arise in a building as a result of heating, ventilating, and air-conditioning (HVAC) systems. These issues can include excessive duct-borne noise, inadequate sound isolation, loud mechanical equipment, and sound paths between spaces created by connected ductwork. In each case, the Trane Acoustics Program (TAP) software can be used to model sound paths and assess the acoustical impact the HVAC system has on a space. This paper will present several case studies representing common mechanical and architectural noise control issues resulting from HVAC system design that were assessed using the TAPsoftware.

TUESDAY AFTERNOON, 24 MAY 2016 — SALON I, 1:00 P.M. TO 3:00 P.M.

Session 2pABa

Animal Bioacoustics and Underwater Acoustics: Cetacean Bioacoustics

Julia Vernon, Chair
Graduate Program in Acoustics, The Pennsylvania State University, State College, PA 16803

Contributed Papers

1:00

2pABa1. What do acoustic tags placed on the back of echolocating dolphins really measure? Whitlow Au (Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, wau@hawaii.edu), James J. Finneran (Space and Naval Warfare Systems Ctr., San Diego, CA), and Brian K. Branstetter (National Marine Mammal Foundation., San Diego, CA)

Suction cup deployed acoustic tags have been used to study the echolocation behavior of a number of odontocetes, yet the relationship between the projected biosonar signals and tag recordings are not known. Acoustic data obtained from these tags consist of the number of clicks emitted, the depth at which the clicks are emitted and the inter-click intervals during a biosonar search. In order to understand the relationship between the emitted signals detected in the front of a dolphin and the signals detected by a tag, a spherical hydrophone was mounted on a wooden model of a tag and mounted on the back of a dolphin via a suction cup. The dolphin was involved in a biosonar discrimination task and the signals were measured 1 m from its blowhole along the acoustic axis of the beam and the acoustic tag placed in five different locations on the animal's back. The signals recorded by the tag were complex with the first pulse being a high frequency resonance-like signal followed by clicks resembling the outgoing clicks but experiencing reflective interference. The peak-to-peak amplitude of the signals measured by the tag was between 40 and 50 dB lower than that of the outgoing signal.

1:15

2pABa2. Does depth matter? Investigating the effect of recording depth on delphinid whistle characteristics and classifier performance. Julie N. Oswald (Bio-Waves, Inc., 364 2nd St., Ste. #3, Encinitas, CA 92024, julie.oswald@bio-waves.net), Marc O. Lammers, Anke Kügler (Oceanwide Sci. Inst., Honolulu, HI), Cory Hom-Weaver, and Robyn Walker (Bio-Waves, Inc, Encinitas, CA)

Seafloor acoustic recorders are commonly used to obtain information about cetaceans, as they allow data to be collected for long periods without

the presence of a human operator. Recordings collected using seafloor instruments do not have associated visual observations, so species must be identified based on their calls. Visually validated acoustic recordings are necessary for training acoustic species classifiers and so most are trained using data collected near the sea surface. The suitability of using classifiers trained using surface recordings to analyze recordings obtained at depth is unknown. To investigate this, we used a vertical array of four Ecological Acoustic Recorders (EARs) spaced 90 m apart to record delphinids at different depths. The same whistles were measured from each EAR and median values of 17 spectrographic variables were compared among EARs for six acoustic encounters. For five of the encounters, there were significant differences in whistle variables among EARs, most commonly in frequency variables. When a random forest classifier was used to identify these whistles to species, the same five encounters were classified as different species when recorded at different depths. These results suggest that caution should be taken when applying classifiers developed using surface data to whistles recorded at depth.

1:30

2pABa3. There must be mucus: Using a lumped-parameter model to simulate the "thump" and "ring" of a bottlenose dolphin echolocation click. Lester Thode (Los Alamos, NM), Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), and Whitlow Au (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, Kaneohe, HI)

Bottlenose dolphin echolocation clicks display a great diversity in temporal and spectral structure, with both unimodal and bimodal spectra observed (Houser *et al*., JASA, 1999). Wavelet scalograms applied to data collected by the Navy Marine Mammal Program and the Bioacoustic Measuring Tool (BMT) (Martin *et al*., JASA, 2005) show that echolocation clicks can display two distinct phases: an initial "thump," followed by an extended "ring" that is adequately modeled by a damped harmonic oscillator. The thump and ring can display either similar or different spectral characteristics, giving rise to a unimodal or bimodal spectrum. A three-mass lumped parameter model, adapted from the speech processing and terrestrial bioacoustics literature, has been used to simulate the oscillation and collision of the dorsal bursae in a dolphin's nasal passage. The three-mass model reproduces many of the time and frequency domain features of entire click trains as well as individual clicks, including unimodal and bimodal spectra. A key insight of the models is that some slight adhesion between the faces of the colliding bursue seems necessary in order to reproduce the high-frequency click structure. A viscoelastic mucus coating could provide one possible mechanism for this required adhesion force. [Data provided by Steve Martin, NMMF.]

1:45

2pABa4. Inter and intra specific variation in echolocation signals among odontocete species in Hawaii, the northwest Atlantic and the temperate Pacific. Tina M. Yack, Kerry Dunleavy, and Julie N. Oswald (Bio-Waves, Inc., 364 2nd St., Ste. #3, Encinitas, CA 92024, tina.yack@bio-waves.net)

Odontocete species use echolocation signals (clicks) to forage and navigate. The aim of this study is to explore inter- and intra-specific variation in clicks among odontocete species in the Northwest Atlantic, Temperate Pacific, and Hawaii. Clicks were examined for seven species of delphinids in the Northwest Atlantic; common dolphin, Risso's dolphin, pilot whale, rough-toothed dolphin, striped dolphin, Atlantic spotted dolphin, and bottlenose dolphin. Newly developed PAMGuard tools were used to automatically measure a suite of click parameters. Five parameters were compared among species; duration, center frequency, peak frequency, sweep rate, and number of zero crossings. Significant differences in duration, center and peak frequency were evident among species within this study area (Dunn's test with Bonferroni adjustment $p < 0.05$). Geographic variation in click parameters among the three study regions was compared for five species; bottlenose dolphin, common dolphin, striped dolphin, pilot whale, and Cuvier's beaked whale. Significant differences in several parameters were found for all species among the regions (Dunn's test with Bonferroni adjustment $p < 0.05$). These results suggest that there are species specific differences in clicks among delphinids and that geographic variation exists for multiple species. The ecological significance of these findings will be discussed along with implications for classifier development.

2:00

2pABa5. Relative abundance of sound scattering organisms in the Northwestern Hawaiian Islands is a driver for some odontocete foragers. Adrienne M. Copeland (Univ. of Hawaii at Manoa, P.O. Box 1106, Kailua, HI 96734, acopelan@hawaii.edu), Whitlow W. Au (Hawaii Inst. of Marine Biology, Kailua, HI), Amanda Bradford, Erin Oleson, and Jeffrey Polovina (Pacific Islands Fisheries Sci. Ctr., NOAA, Honolulu, HI)

Previous studies in the Northwestern Hawaiian Islands (NWHI) focused on shallower communities in and near reefs and did not investigate the organisms living in deeper waters that some apex predators rely on for food, e.g., some odontocetes forage at depths greater than 400 m. To examine the relationship between deep-diving odontocete predators and prey, a Simrad EK60 echosounder operating at 70 kHz collected acoustic abundance throughout the NWHI from May 7 to June 4, 2013. Visual and passive acoustic surveys for marine mammal presence were conducted concurrently with the echosounder. Two broad scattering layers were found, a deep layer from 325 to 670 m and a shallow layer from 0 to 195 m. The highest densities of both deep and shallow scattering organisms were associated with deep slopes of banks and atolls. Beaked and short-finned pilot whale sightings occurred in locations of high scattering density associated with slopes of atolls and banks. It is hypothesized that the high scattering organisms associated with these features are similar to the mesopelagic boundary community found in the Main Hawaiian Islands and support a food web representing the prey of the cetaceans.

2:15

2pABa6. Beaked whale acoustic versus visual detection. Odile Gerard (DGA Naval Systems, Ave. de la Tour Royale, Toulon 83000, France, odigea@gmail.com)

Because of their sensitivity to anthropogenic noise, research on beaked whale habitat is particularly important. During 2010 and 2011, NATO Undersea Research Centre (NURC) conducted sea trials dedicated to marine mammals, in areas of potential beaked whale habitat. The first one took place in North Eastern Atlantic Ocean, Southwest of Portugal, and the second one took place in the Gulf of Genoa, Mediterranean Sea. For both trials: weather conditions allowing, during daylight there were two teams of visual observers, working in two shifts, scanning the horizon and taking note of marine mammal encounters. Acoustic data were collected with the CPAM (Compact Passive Acoustic Monitoring), designed by NURC. The total usable bandwidth is up to 80 kHz. The CPAM was deployed at a depth between 100 and 200 m for about 20 h a day Beaked whale detection obtained by visual observers and by passive acoustic are analyzed. The number of detections and the information obtained by each method are compared. The advantages and drawbacks are highlighted.

2:30

2pABa7. Seasonal variability in distribution of fin whales around Wake Island. Julia A. Vernon and Jennifer L. Miksis-Olds (Graduate Program in Acoust., Appl. Res. Lab, The Penn State Univ., State College, PA 16803, jav232@psu.edu)

Passive acoustic monitoring in population density estimation of marine mammals provides an efficient and cost-effective alternative to visual surveys. However, one challenge that arises with this method is uncertainty in the animal distribution. Information about distribution is needed in order to account for spatial variability in the probability of detection. Consideration also needs to be given as to how distribution varies between seasons, as seasonal variability also needs to be incorporated into the density estimation. This paper presents bearing estimates of fin whales around Wake Island in the Equatorial Pacific Ocean, using low-frequency ambient noise data (5–115 Hz) acquired by the Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO) International Monitoring System. Bearings were initially calculated using time delay information from the cross-correlation of received signals. However, a simple cross-correlation is not a viable option for many calls, due to distortion of the waveform as a result of modal dispersion, and alternate methods of determining time delays of received signals are discussed. Bearings were calculated for individuals detected over a period of three years: May 2007 to May 2010. Seasonal variability in distribution is presented. [This work was supported by the Office of Naval Research.]

2:45

2pABa8. Long-term monitoring of Physeteroidea (sperm whales, dwarf, and pygmy sperm whales) in the Central and Western Pacific. Karlina Merkens (CRP, NOAA/PIFCS (Lynker Tech.), 3710 SW Caldew St., Portland, OR 97219, kmerkens@gmail.com), Anne Simonis (UCSD/SIO, La Jolla, CA), and Erin Oleson (CRP, NOAA/PIFSC, Honolulu, Hawaii)

The superfamily Physeteroidea includes three extant species: the sperm whale (*Physeter macrocephalus*), the dwarf sperm whale (*Kogia sima*), and the pygmy sperm whale (*K. breviceps*). Despite extreme difference in size between the *Kogia* spp. and their large *Physeter* relative, all three share ecological and acoustic traits relating to their deep-diving behavior and high rates of acoustic activity. All three species can be found across the Central and Western Pacific ocean, an area that has been monitored using passive acoustics (High-frequency Acoustic Recording Packages, HARPs) for more than 10 years. We identified sperm whale and *Kogia* spp. signals in the long-term HARP records from 13 locations across the Central and Western Pacific ocean. A combination of automated tools and human analysis were used to record detection events of both types of signals. While sperm whales were found at all 13 locations, the *Kogia* species (which cannot yet be distinguished acoustically) were detected at approximately half of the sites. Presence of sperm whale signals was modeled to determine if temporal parameters, such as lunar cycle and day of the year, could explain patterns of presence. Across the whole region the best model included the day of the year and the recording site, while sub-regions and site-specific models had slightly different combinations of parameters.

TUESDAY AFTERNOON, 24 MAY 2016 — SALON I, 3:30 P.M. TO 5:00 P.M.

Session 2pABb

Animal Bioacoustics: Animal Bioacoustics Poster Session

Benjamin N. Taft, Chair
Landmark Acoustics LLC, 1301 Cleveland Ave., Racine, WI 53405

All posters will be on display and all authors will be at their posters from 3:30 p.m. to 5:00 p.m.

Contributed Papers

2pABb1. The spatial unmasking of pure tones by laboratory mice. Laurel A. Screven (Psych., Univ. at Buffalo, B29 Park Hall, Buffalo, NY 14260, laurelsc@buffalo.edu)

The ability of animals to identify signals in their acoustical environment when confronted with sounds from multiple sources relies on them extracting the important information and filtering out background noise. Previous reports have shown that many animals are able to attend to a single sound more easily when that signal is spatially separated from the background noise. This spatial unmasking of a sound has previously not been behaviorally measured in mice, despite their use as a model for human hearing and communication. The present experiment examined if laboratory mice are able to show spatial release from masking, as seen in other animals, by testing the detection of 2, 4, 8, 12, 16, and 24 kHz pure tones in the presence of a white noise masker. The masker was spatially coincident with the signal or separated by 90°. We hypothesized that the mice would experience more spatial unmasking for the higher frequency tones since they rely heavily on these frequency ranges for their communication signals. Preliminarily, we have found that mice are able experience spatial unmasking at some, but not at all frequencies.

2pABb2. Intensity difference limens in mice. Anastasiya Kobrina, Katrina Toal, Kali Burke, and Micheal L. Dent (Psych., SUNY Univ. at Buffalo, B23 Park Hall, Amherst, NY 14261, akobrina@buffalo.edu)

The ability to distinguish between sounds of differing intensities is a universal auditory process. Previous researchers have examined the intensity discrimination abilities of goldfish, parakeets, and several mammals including feral and house mice using behavioral approaches. CBA/CaJ mice are frequently used as a model for human hearing research, yet intensity difference limens (IDLs) have not been measured in this strain. The aim of the present study was to establish IDLs for 12, 16, 24, and 42 kHz tones at 10 and 30 dB SL. Eight adult animals were trained and tested in a discrimination task using operant conditioning procedures with positive reinforcement. IDLs were variable across all frequencies for 10 dB SL, but remained stable for 12, 16, and 42 kHz at 30 dB SL, with elevated IDLs at 24 kHz. Calculations of the Weber fractions at each sensation level tested showed that Weber's law does not hold true for IDLs in CBA/CaJ mice. Overall, thresholds were comparable to those found in other animals.

2pABb3. Utilizing the sound of predators to cast out wild animals. Myungsook Kim (English, Soongsil Univ., Sangdo-ro 369, Seoul 06978, South Korea, kimm@ssu.ac.kr), Ik-Soo Ahn, and Myung-Jin Bae (Information and TeleCommun., Soongsil Univ., Seoul, South Korea)

Recently, there have been many reports of damages caused by animals such as wild pigs and wild dogs in the rural farms of Korea. Damages were extensively widespread from agricultural crop and produce to domestic animals. To efficiently cast out wild animals, the recorded sound of predators such as lions and tigers was used. The speakers were installed around the farms and broadcasted the sound of predators every 3 h. Although the damages were a little bit reduced after broadcasting, it did not complete its purpose and still some damages were reported. We analyzed the recording quality of the sound and found out some deficiencies in it, the lack of low frequency sound below 100 Hz in particular. In order to improve its sound quality, we installed the speakers inside of water pipes with 1 m in diameter. As a result, we earned 90% in the index of similarity between the original crying sound and the broadcasted crying sound of predators in its sound spectra. Since the improved sound recording was broadcasted for 1 min every 3 h in the farms, the damages have not been reported.

2pABb4. Acoustical and vibratory characteristics of a synthetic mallard syrinx replica. Taylor B. Groom (Mech. Eng., Brigham Young Univ. Idaho, Rexburg, ID 83460, gro11017@byui.edu), Tobias Riede (Physiol., Midwestern Univ., Glendale, AZ), and Scott L. Thomson (Mech. Eng., Brigham Young Univ. Idaho, Rexburg, ID)

The study of the physics of human voice production has benefited by the use of synthetic human vocal fold replicas. Extending this concept to avian vocalization, a functioning model of a male mallard syrinx for studying the relationship between syrinx anatomical features and vocalization characteristics is described here. The mallard syrinx is characterized by two sets of labia located at the tracheobronchial juncture, and vocalization is produced by the flow-induced vibration of these labia. Shape and histological composition of labia demonstrate similarities to mammalian vocal folds. In this study, life-sized synthetic replicas based on CT data of an adult male mallard syrinx, including airway and labia regions, are studied. The replicas are fabricated using exceedingly flexible silicone materials that have been previously used in synthetic human vocal fold replicas. The replicas are mounted in tube-like airways representing the trachea and bronchi, with flow generated by an adjustable air supply. In this presentation, the CT-based syrinx model, including fabrication process, will be introduced. The model response, including acoustical output and vibratory motion, will be described and compared to previous models and to existing data about mallard vocalization.

2pABb5. Neyman-Pearson detection of underwater bioacoustic signals. Florian Dadouchi (Gipsa-Lab, Univ. Grenoble Alpes, 11 rue des Mathématiques, Grenoble Campus, SAINT MARTIN D'HERES BP46, F-38402 CEDEX, France, florian.dadouchi@gipsa-lab.fr), Julien Huillery (Méthodes pour l'ingénierie des systèmes, Laboratoire Ampère (CNRS UMR5005)., Ecully, France), Cédric Gervaise (Chorus Foundation., Grenoble Cedex 1, France), and Jérôme I. Mars (Gipsa-Lab, Univ. Grenoble Alpes, Saint Martin d'Hères, France)

The use of passive acoustic for the classification, localization, and density estimation of populations of marine mammals is a current area of interest. It is a cheap and an efficient alternative to visual surveys. However, the lack of an efficient automatic detector for unknown marine mammal calls greatly undermines the feasibility of those tasks, especially when dealing with species showing a great variability of calls. This study adds one more step toward the fully automatic detection of unknown bioacoustic signals in impulsive, non-stationary, and colored ocean noise. The detection procedure is a two-steps fully statistical method solely based on the knowledge of the background noise in the spectrogram. The first step models the noise power as a chi-squared distribution, which parameter is estimated. The signal is then detected using a Neyman-Pearson approach, providing a binary spectrogram that contains false and true detections. The second step removes a significant amount of false detections from the binary spectrogram. The time-frequency distribution of false detections is fitted with a correlated binomial distribution, which is used to discriminate patches of detections (signal) from uniformly distributed detections (false alarms). Examples showing the applicability of this method on several real underwater sounds are presented.

Session 2pBA

Biomedical Acoustics: Biomedical Acoustics Student Paper Competition (Poster Session)

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

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

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

2aBAa1. Toward validation of shear wave elastography using vibration rheometry in soft gels
Student author: Sanjay Yengul

2aBAb1. Jet formation of contrast microbubbles in the vicinity of a vessel wall
Student author: Nima Mobadersany

2aBAb4. Tracking kidney stones during shock wave lithotripsy
Student author: Kya Shoar

2aBAb5. Effects of acoustic parameters on nanodroplet vaporization
Student author: Krishna Kumar

2aBAb6. Effects of ultrasound in presence of microbubbles on cartilage tissue regeneration in three-dimensional printed scaffolds
Student author: Mitra Aliabouzar

2aBAb7. Microbubble response to dual frequency excitation for broadband contrast imaging
Student author: Christina Keravnou

2aBAb8. Lytic efficacy of tissue plasminogen activator and ultrasound in porcine clots doped with barium sulfate in vitro
Student author: Shenwen Huang

3aBA7. Ultrasound-mediated transport of nanoparticles and the influence of particle density
Student author: Harriet Lea-Banks

3pBA6. Phased array techniques for multiple focus synthesis in transcranial focused ultrasound
Student author: Alec Hughes

4pBA1. Simulation and implication of a two-dimensional phased array flexible ultrasound system for tissue characterization
Student author: Zhihua Gan

4pBA3. Development of a nonlinear model for the pressure dependent attenuation and sound speed in a bubbly liquid and its experimental validation
Student author: Amin Jafari Sojahrood

4pBA5. Photoacoustic imaging of muscle oxygenation during exercise
Student author: Clayton Baker

Session 2pEA

Engineering Acoustics: General Topics in Engineering Acoustics II

Kenneth M. Walsh, Chair
K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Chair's Introduction—1:10

Contributed Papers

1:15

2pEA1. Sensitivity analysis of the performance of a linear alternator at mechanical resonance under thermoacoustic power conversion conditions. A. H. Ibrahim (American Univ. in Cairo, Egypt 11835, Egypt, abdelmaged@aucegypt.edu), Ahmed Yassin, and Ehab Abdel-Rahman (American Univ. in Cairo, New Cairo, Egypt)

A linear alternator lie at the interface between the acoustic power generated by a thermoacoustic engine and an electric load, and thus, its performance is significantly controlled by its matching with the thermoacoustic engine and with the electric load. Under thermoacoustic power conversion conditions, small unavoidable changes in the operating conditions may incur significant effects on the alternator performance. In this work, the sensitivities of several alternator performance indices, namely, the acoustic-to-electric conversion efficiency, the mechanical stroke, and some of the main alternator losses (mechanical damping loss, seal loss, and electric copper loss), are examined to small changes in four operating conditions. Using the methodologies of the design of experiments and sensitivity analysis, a scheme of experiments is designed and carried-out to analyze the sensitivity of these indices to $\pm$ 10% changes in operating conditions at mechanical resonance. The operating conditions considered are the gas mixture composition, the mean gas pressure, the dynamic pressure ratio acting on the alternator and the electric load. The results reveal how variations in each of these operating variables as well as variations in their combined interactions affect the alternator's performance with respect to the results obtained in a reference experiment.

1:30

2pEA2. Performance of the phase and amplitude gradient estimator intensity method in interference fields. Darren K. Torrie, Eric B. Whiting, Michael T. Rose, Kent L. Gee, and Scott D. Sommerfeldt (Phys., Brigham Young Univ., Eyring Sci. Ctr., Provo, UT 84606, darren@torriefamily.org)

The phase and gradient amplitude estimator (PAGE) method improves the frequency bandwidth of estimated acoustic intensity over the traditional p-p method without altering the spacing between microphones [D. C. Thomas *et al*., J. Acoust. Soc. Am. **137**, 3366–3376 (2015)]. For many broadband sources, accurate estimates may be obtained beyond the spatial Nyquist frequency by unwrapping the phase of a transfer function used in obtaining the phase gradient. However, inaccurate phase unwrapping in interference fields, such as those produced by two loudspeakers with equal strengths but opposite phase, has been observed. This results in erroneous intensity vectors. A two-dimensional, multi-microphone intensity probe was employed to investigate this phenomenon. Findings include: (a) the unwrapping error does not occur for all interference nulls, but is more likely to occur for deeper interference nulls where there is reduction in coherence; (b) rotation of the probe in the field alters which pair-wise transfer function unwraps erroneously, but does not significantly alter the direction that contains the inaccurate intensity vectors; (c) removal of the interference nulls by driving the loudspeakers as incoherent sources allows for phase unwrapping to occur multiple times and accurate estimation of intensity vectors beyond 10 kHz. [Work supported by NSF.]

1:45

2pEA3. Comparison of pressure-based intensity measurements for different probe designs and estimation methods. Michael T. Rose, Darren K. Torrie, Reese D. Rasband, Kent L. Gee, and Scott D. Sommerfeldt (Dept. of Phys., Brigham Young Univ., Provo, UT 84602, rosemission@surewest.net)

Acoustic intensity measurements have traditionally used cross spectral methods with multimicrophone probes to estimate the required pressure and particle velocity. The phase and gradient estimator (PAGE) method increases probe bandwidth without modifying microphone spacing as compared to the traditional cross spectral method [D. C. Thomas *et al*., J. Acoust. Soc. Am. **137**, 3366–3376 (2015)]. In this study, high-frequency probe bandwidth obtained using both the PAGE method and the traditional method is compared across three different multimicrophone probe designs: a three-axis, six-microphone probe with variable solid spacer, a three-axis, four-microphone spherical probe, and a two-axis, four-microphone probe developed in-house for rocket measurements. Broadband, anechoic loudspeaker measurements are used to examine the performance of each probe and estimation method. These include a single 7.6 cm diameter, enclosed loudspeaker driver, and two drivers of equal strength but opposite phase. Intensity measurements in a grid of 41 $\times$ 41 points inside a one square meter area are presented. Probe design robustness is determined by considering bias errors in calculation methods and probe scattering. [Work supported by NSF.]

2:00

2pEA4. Adaptive imaging processing for reducing ultrasonic ringdown interference pattern. Zhijuan Zhang, Douglas Patterson, Roger Steinsiek, and Wei Han (Baker Hughes, 2001 Rankin Rd., Houston, TX 77073, Zhijuan.Zhang@bakerhughes.com)

In the oil and gas industry, ultrasonic imaging tools are actively used in stress identification and fracture detection. Pulse-echo transducers measure the properties of the echoes reflected from the formation, and the signal peak amplitude and travel time are typically displayed as a circumferential image of the borehole wall. A well-known problem is that the pulse-echo transducers have internal ringdown which constantly modulates formation echoes. As a result, the acquired image is often dominated by a "woodgrain" interference pattern which degrades image quality. Traditional ringdown reduction methods try to remove the ringdown effect by estimating the ringdown waveform. However, the ringdown signal varies versus borehole conditions, which make it inadequate for traditional methods to efficiently reduce the image interference pattern. In this paper, we proposed an image processing method which significantly reduces the interference pattern. This new image processing method does not require any ringdown waveforms. It adaptively estimates the ringdown modulation template and after that constructs the interference amplitude pattern. Examples shows promising results and have proven the propose method is more efficient than traditional ringdown reduction methods on the ringdown interference pattern reduction.

2:15

2pEA5. *In-situ* ultrasonic measurements of creep specimens. Manton J. Guers (Structural Acoust., Penn State Appl. Res. Lab., PO Box 30, State College, PA 16804, mjg244@psu.edu) and Bernhard R. Tittmann (Eng. Sci. & Mech., Penn State Univ., University Park, PA)

Performing *in-situ* measurements of specimens in research reactors is challenging because of the environmental conditions. In this work, ultrasonic guided waves were investigated for performing *in-situ* measurements of the change in length of creep specimens. Both theoretical calculations and experimental measurements were used to determine the proposed method's sensitivity to changes in temperature and elongation. The experimental tests demonstrated that careful consideration must be given to the signal processing of the data. Successful measurements of the creep elongation of a 3 in. gauge length specimen were demonstrated.

2:30–2:45 Break

2:45

2pEA6. Active compensation of nonlinear distortion in balanced armature receivers. Buye Xu and Tao Zhang (Signal Processing Res., Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344, buye_xu@starkey.com)

Balanced armature receivers have been widely used in hearing aids due to their higher efficiency and more discrete size comparing to moving coil loudspeakers. For the application of hearing aids, it may require a very high output sound pressure level (up to 110 dB in an opened ear canal) and, yet, minimum nonlinear distortions. This requirement often results in a larger receiver in size, which is less comfortable for wearing. It is highly desired if the linear range of a smaller receiver can be broaden without increasing its physical size. Active compensation of nonlinearity for moving coil loudspeakers through signal processing means have been well studied. However, similar research for balanced armature receivers is rarely seen in the literature. Although the two types of transducers differ in many aspects, they share commonalities in the nonlinearity mechanism. The current study surveys the active compensation techniques developed for moving coil loudspeakers and explores the possibility of adopting them to reduce the nonlinearity of balanced armature receivers.

3:00

2pEA7. Size differentiation of a continuous stream of particles using acoustic emissions. Ejay Nsugbe, Andrew Starr, Peter Foote, Cristobal Ruiz-Carcel, and Ian K. Jennions (Cranfield Univ., 79 Lower Shelton Rd., Marston Moretaine MK43 0LN, United Kingdom, e.nsugbe@cranfield.ac.uk)

Procter and Gamble (P&G) requires an online system that can monitor the particle size distribution of their washing powder mixing process. This would enable the process to take a closed loop form which would enable process optimization to take place in real time. Acoustic emission (AE) was selected as the sensing method due to its non-invasive nature and primary sensitivity to frequencies which particle events emanate. This work details the results of the first experiment carried out in this research project. The first experiment involved the use of AE to distinguish sieved particle which ranged from 53 to 250 microns and were dispensed on a target plate using a funnel. By conducting a threshold analysis of the peaks in the signal, the sizes of the particles could be distinguished and a signal feature was found which could be directly linked to the sizes of the particles.

3:15

2pEA8. Recording anechoic gunshot waveforms of several firearms at 500 kilohertz sampling rate. Tushar K. Routh and Robert C. MAHER (Dept. of Elec. and Comput. Eng., Montana State Univ., 610 Cobleigh Hall, Bozeman, MT 59717, tushar.routh@msu.montana.edu)

Acoustic gunshot signals consist of a high amplitude and short duration impulsive sound known as the muzzle blast. This experiment involved documenting gunshot muzzle blast sounds produced by eight commonly used firearms. An elevated bracket (3 m above the ground) was built to achieve a quasi-anechoic environment for the duration of the muzzle blast. Twelve microphones (GRAS 46DP) were mounted on the bracket in a semi-circular arc to observe the azimuthal variation of the muzzle blast. Signals were recorded using LabVIEW. Similarities and differences among waveforms are presented.

3:30

2pEA9. Vorticity-involved acoustic damping performance of an in-duct orifice with a bias flow. Dan Zhao, Nuomin Han (School of Mech. and Aerosp. Eng., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, Singapore 639798, Singapore, zhaodan@ntu.edu.sg), and Xiao Jin (School of Energy and Power Eng., Jiangsu Univ. of Sci. and Technol., Zhenjiang City, Jiangsu, China)

Acoustic damping performance of an in-duct perforated orifice with a bias flow in terms of power absorption and reflection coefficients are evaluated in this work. For this, experimental measurements of a cold-flow pipe system with a diameter of $2b$ with an in-duct perforated plate implemented are conducted first. It is shown that the maximum power absorption Δ_{max} and reflection coefficients χ_{max} are approximately 80% and 90%, respectively. In addition, Δ and χ are periodically changed with the forcing frequency. To simulate the experiments and gain insights on the damping performance of the orifice with a diameter of $2a$, a 1D theoretical model embodying vorticity-involved damping mechanism is developed. It is based on the modified form of the Cummings equation describing unsteady flow through an orifice and the Cargill equation describing acoustically open boundary condition at the end of the downstream duct. It is shown that Δ and χ are strongly related to (1) the mean flow Mach number, (2) forcing frequency ω, and (3) porosity $\eta = a/b$, and (4) the downstream pipe length L_d. Theoretical predictions are found to agree well with experimental measurements. This confirms that the model has the potential to predict the acoustic damping performance of in-duct orifices.

3:45

2pEA10. Absorption of axial plane waves by using double- and single-layer perforated liners. Dan Zhao, Nuomin Han (School of Mech. and Aerosp. Eng., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, Singapore 639798, Singapore, zhaodan@ntu.edu.sg), and Linus Ang (Dept. of Mech. Eng., National Univ. of Singapore, Singapore, Singapore)

In this work, the acoustic damping performances of multiple double- and single-layer perforated liners are experimentally and numerically evaluated. These acoustic liners are associated with different open-area-ratios, thus enable the porosity effect to be studied. Controllable joint bias-grazing flow is applied to the liners. This simulates the flow configurations of a real engine system involving both bias and grazing flow. The damping performance of these liners is characterized by power absorption or transmission loss coefficient. Such coefficient is measured, as the tonal noise frequency is changed from 200 to 800 Hz. It is shown that the grazing flow can reduce the maximum power absorption coefficient, while the bias flow can improve the liners damping performance. The double-layer liner is associated with a larger maximum power absorption and a broader frequency range in comparison with that of the single-layer one. Finally, it is found that when the open area ratios of the inner and outer liners are approximately 1.1%, the damping performances of these liners are dramatically reduced, especially at higher frequency range. Increasing the open area ratios of the inner or outer liner leads to the maximum power absorption and the effective frequency range being dramatically improved.

Session 2pMU

Musical Acoustics: Pitch, Dynamics, and Vowel Tuning in Choral Voice

Ingo R. Titze, Chair
National Center for Voice and Speech, 136 South Main Street, Suite 320, Salt Lake City, UT 84101-3306

Chair's Introduction—1:00

Invited Papers

1:05

2pMU1. Maximizing the acoustic benefits of vocal vibration and resonance in choral singing. Laurier A. Fagnan (Campus Saint-Jean, Univ. of AB, 4915 - 114 B St., Edmonton, AB T6H 3N2, Canada, Lfagnan@ualberta.ca)

As an artistic discipline, choral singing often shies away from fully exploiting the vocal and acoustic benefits of vibration and resonance. It is sometimes thought that those two properties are more germane to the solo voice than to ensemble singing where "sameness" is often lauded above all else and *blend* can easily turn to *bland*. As Berton Coffin aptly states, "there is no reason to have a Stradivarius sound like a cigar-box violin so that both will sound the same." This session will show how the Stradivarius, complete-timbre-energy approach espoused by the old masters of the *bel canto* manner of singing is appropriate and beneficial to the art of choral singing. It will highlight the many acoustic advantages a choir can enjoy by applying the principles of regular, focused vibration and balanced, *chiaroscuro* (bright-dark) resonance throughout the voice's range and dynamic extent. Analyses such as high-speed videostroboscopy and real-time spectrography will help to better define and bridge the scientific and artistic aspects of energized choral sound. It will be shown that the proper application of *bel canto* principles has a positive effect on spectral balance, notably the presence of Singers' Formant energy, as well as on intonation and text intelligibility.

1:35

2pMU2. Pitch, dynamics, and vowel tuning in choral voice: Utilizing resonance strategies to train stylistic variance for choral singers. Jonathan D. Harris and Laurel Mehaffey (Music, College of the Holy Cross, 1 College St #195A, Worcester, MA 01610, jdharris@holycross.edu)

Building off of the notion of absolute timbre and its influence on the perceptual and acoustical definition of vowel, we explore acoustic resonance strategies that can help singers address the challenges of performing in varied styles. Choral singers have to manage variation in singing styles more than any other musician, and yet, they are often the musicians with the least amount of formal training. Most choral directors choose a one-size-fits-all resonance strategy that accentuates the fundamental and suppresses other harmonics. This "Choral Cathedral" resonance strategy minimizes the challenges inherent in balancing voices whose power spectra vary, but eliminates many of the possibilities available to the voice. By training singers to recognize resonant strategies, and how they function and feel, singers can begin to explore more of their voice's potential. Ultimately, this creates a more satisfying artistic experience for singer and audience alike. We use spectral analysis of different resonant strategies to show the impact of the absolute timbre of individual harmonics. We show how the vocal tract, even beyond the first and second formant, can create predictable acoustical environments unique to certain singing styles that singers can learn to achieve.

2:05

2pMU3. Loudness range of a choir based on choir size and voice range profiles of individuals. Ingo R. Titze (National Ctr. for Voice and Speech, Univ. of Utah, 156 South Main St., Ste. 320, Salt Lake City, UT 84101-3306, ingo.titze@utah.edu)

The size of a choir is based on many factors, including availability of singers, size of the performance hall, economics of practice and performance, and the desired ensemble sound. It is generally assumed that a large choir has a greater dynamic range than a small choir. It is also generally assumed that a choir with well-trained singers can produce a greater dynamic range. A theoretical study is presented here that combines sound pressure level and loudness of homogeneous and non-homogeneous groups of singers to produce an overall sound level profile of a choir. Internal constraints are self-to-other ratio and individual voice range profiles of choir members. Results indicate that the dynamic range of a choir is determined mostly by the dynamic range of individuals with wide ranges, assuming that inhomogeneity is allowed. Choir size makes little difference. Singers with small dynamic ranges have little effect on choir dynamics unless louder voices are selectively turned off in pianissimo passages.

2:35

2pMU4. Pitch control in professional and amateur singers. Pasquale Bottalico (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Lansing, MI 48910, pb@msu.edu), Simone Graetzer, and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

One aspect of major relevance to singing is the control of fundamental frequency. Ten amateur and ten professional singers sang two-octave arpeggi with three different levels of external auditory feedback, two tempi and two articulations (legato or staccato). The effects on pitch inaccuracy (defined as the distance in cents between the reference note and the sung note) of the following conditions were evaluated: (1) level of external feedback, (2) tempo (slow or fast), (3) articulation (legato or staccato), (4) tessitura (low, medium, or high), and (5) semi-phrase direction (ascending or descending). It was observed that inaccuracy was greatest in the descending semi-phrase arpeggi produced at a fast tempo and with a staccato articulation, especially for non-professional singers. The magnitude of inaccuracy was also relatively large in the high tessitura relative to the low and medium tessitura for such singers. Counter to predictions, when external auditory feedback was strongly attenuated or canceled by the hearing protectors, non-professional singers showed greater pitch accuracy than in the other external feedback conditions, which indicates the importance of internal auditory feedback in pitch control. With an increase in training, the singer's pitch inaccuracy decreases.

2:50

2pMU5. The effect of *chiaroscuro* and *coup de glotte* training on reducing the effects of intrinsic pitch in sung vowel transitions. Grace H. Crosby, Benjamin V. Tucker, and Laurier Fagnan (Linguist, Univ. of AB, 116 St. & 85 Ave., Edmonton, AB T6G 2R3, Canada, gcrosby@ualberta.ca)

Intrinsic pitch is an acoustic value that differs depending on vowel height. Sapir (1989) describes the various theories that provide an explanation for intrinsic pitch. Essentially, vowels which are produced higher in the filter (e.g., /i/ and /u/) tend to be pitched slightly higher than vowels produced at lower locations in the filter (e.g., /a/). The purpose of this study is to determine whether the *chiaroscuro* and *coup de glotte* vocal techniques used in the *bel canto* method of singing reduce the natural effects of intrinsic pitch of sung vowels by attempting to equalize both resonance and vibrational properties across the vowel spectrum. Six singers (2 sopranos, 2 tenors, and 2 basses) were recorded in this study. Participants sang a set of vowel transition exercises in a pre-training setting, then were trained in *chiaroscuro* and *coup de glotte* techniques, after which they did a post-training recording. Pitch, formant values, and amplitude (as mean values and at various times within each vowel) were extracted for a before versus after comparison. The results will be discussed in relation to how this technique might be helpful in reducing the effect of intrinsic pitch on sung vowels.

3:05

2pMU6. How singing experience affects the "brightness" of humming? Ae Na Yang, Marina Takabayashi, Sachi Itagaki, Hayato Kikuchi, and Kota Kobayasi (Doshisha Univ., Kyoto, Kyotanabe-City, Kyoto Tatara Miyako Tani 1-3, Japan, bmm1121@mail4.doshisha.ac.jp)

During singing practice, people are often told to sing in bright voice to improve singing performance. The purpose of this study is to determine how singing practice affects the acoustical characteristics and expressiveness of "brightness." In this study, eight singers (having experience in chorus for 0 to 13 years) sang three types of humming with G4 (392 Hz). Singers were told to sing either "bright," "dark," or "normal" humming. Subjects ($n = 16$) listened to different types of humming in pairs, and judged which was brighter. The results showed that the subjects perceived "bright" humming as bright. As singers practiced longer, their "brightness" was more expressive. Acoustical analysis revealed that fundamental frequencies of "bright" humming were higher than dark humming by 3 Hz, and that bright humming had higher amplitude in spectrum from 4900 to 6400 Hz by 4.5 to 6.5 dB. In the following experiment, subjects listened to and judged the brightness of humming, whose amplitude in the range was systematically amplified. As a result, the more amplified the humming had in the range, the more likely subjects evaluated it as bright. Our results suggest that singers can obtain the ability to express "brightness" by producing these acoustical characteristics through practicing singing.

Session 2pNS

Noise and Signal Processing in Acoustics: Statistical Learning Techniques in Noise Research

Jonathan Rathsam, Cochair
NASA Langley Research Center, MS 463, Hampton, VA 23681

Edward T. Nykaza, Cochair
ERDC, 2902 Newmark Drive, Champaign, IL 61822

Chair's Introduction—1:30

Invited Papers

1:35

2pNS1. Deep learning for unsupervised feature extraction in audio signals: A pedagogical approach to understanding how hidden layers recreate, separate, and classify audio signals. Edward T. Nykaza (ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61822, edward.t.nykaza@usace.army.mil), Arnold P. Boedihardjo (ERDC-GRL, Alexandria, VA), Zhiguang Wang, Tim Oates (Comput. Sci. and Elec. Eng., UMBC, Baltimore, MD), Anton Netchaev, Steven L. Bunkley (ERDC-ITL, Vicksburg, MS), and Matthew G. Blevins (ERDC-CERL, Champaign, IL)

Deep learning is becoming ubiquitous; it is the underlying and driving force behind many technologies we use everyday (e.g., search engines, fraud detection warning systems, and social-media facial recognition algorithms). Over the past few years, there has been a steady increase in the number of audio and acoustics related applications of deep learning. But what is exactly going on under the hood? In this paper, we focus on deep learning algorithms for unsupervised feature learning. We take a pedagogical approach to understanding how the hidden layers recreate, separate, and classify audio signals. We begin with a simple pure tone dataset, and systematically increase the complexity of this dataset in both frequency and time. We end the presentation with some feature extraction examples from real-world environmental recordings, and find that these features are easier to interpret given the understanding developed from the simpler tone datasets. The unsupervised feature learning techniques explored in this paper include: restricted Boltzmann machines (RBMs) and auto-encoders (AEs).

1:55

2pNS2. The importance of feature selection in supervised machine learning problems in acoustics. Eric A. Dieckman (Mech. Eng., Univ. of New Haven, 300 Boston Post Rd., West Haven, CT 06516, edieckman@newhaven.edu)

As machine learning processes begin to gain traction and are applied to problems in acoustics, there is a need to better understand the importance of all stages of the process. Here, we discuss the preprocessing stage, where features are extracted from raw data and the best features are chosen using algorithms such as linear discriminant analysis (LDA). This allows the creation of a small-dimensional, information-dense feature vector to be used in the machine learning process. Applications discussed include classification of ground vehicles based on their acoustic backscatter signature and classification of wall structures based on transmission loss measurements.

2:15

2pNS3. Machine-learning models for the prediction of long-range outdoor sound propagation. Carl R. Hart, D. K. 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)

Long-range outdoor sound propagation is characterized by a large variance in sound pressure levels due to factors such as refractive gradients, turbulence, and topographic variations. While conventional numerical methods for long-range propagation address these phenomena, they are costly in computational memory and time. In contrast, machine-learning algorithms provide very fast predictions, which this study considers. Observations from either experimental data, or surrogate data from a numerical method, are required for the training of machine-learning models. In this study, a comprehensive training set for the machine learning was created from excess attenuation predictions made with a Crank-Nicholson parabolic equation (CNPE) model. Latin hypercube sampling of the parameter space (source frequency, meteorological factors, boundary conditions, and propagation geometries) generates a set of input for the CNPE model and machine-learning models. Consideration is given to ensemble decision trees, ensemble neural networks, and cluster-weighted models for nonlinear regression. The large variance in excess attenuation, from the CNPE model, presents a challenge for accurate machine-learning model predictions. For example, given 5000 samples the overall root-mean-square error for an ensemble decision tree model is 6.7 dB. Errors related to sample size, modeling approaches, and propagation ranges are quantified in this study.

2:35

2pNS4. On the distribution of impulsive sound events for environmental noise assessment. Frank Van den Berg and Frits Van der Eerden (TNO, Oude Waalsdorperweg 63, The Hague 2597 AK, Netherlands, frank.vandenberg@tno.nl)

The noise levels of impulsive sounds are subject to variation, mostly due to changes in the meteorological situation which have a strong influence on the noise propagation. For environmental noise assessment studies the variation in the single events levels as well as the long term (averaged) level should be considered. To calculate these one can use a number of variations for the atmospheric absorption and the excess attenuation, alongside with histograms of for instance the wind speed and wind directions. Just recently, a new approach has been enforced in the Netherlands to assess the sound from shooting ranges. Details on this calculation method will be given in which meteorological classes are used to account for varying wind speed and temperature gradients. This method makes it also possible to describe the distribution of occurring sound exposure levels around shooting ranges. The distribution of impulsive sounds is further illustrated with data obtained from projects carried out on: long range propagation of high-energy blasts, monitoring military training areas with detonations and muzzle blast noise, and monitoring fireworks in an urban area.

2:55–3:10 Break

3:10

2pNS5. Trans-dimensional Bayesian approaches to room acoustic modal analysis. Douglas Beaton and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, beatod@rpi.edu)

In noise control engineering, it is often desirable to assess the modal behavior of mechanical structures or a room and determine what implications any modes present in the system under investigation will have on noise. The application of Bayesian methods to analyze modal behavior in experimentally measured room impulse responses is of main interest in the current work. These methods extract relevant information from the room impulse response in the time domain and can be used to estimate modal parameters such as mode frequency, amplitude, decay time, and phase. Previous efforts in Bayesian modal analysis divided the analysis into two distinct steps: model selection (determining the appropriate number of modes), and parameter estimation (determining the parameters of each mode). This work considers approaches that combine the two steps into a single trans-dimensional operation. The number of modes in a given model is defined as a parameter of that model. By doing so, the task of model selection becomes part of the parameter estimation, which itself becomes a trans-dimensional problem. Approaches to solve this trans-dimensional problem are applied to both simulated and experimentally measured room impulse responses. Results are compared with other Bayesian approaches and also with conventional Fourier analysis.

3:30

2pNS6. Statistical analysis of multilayer porous absorbers with Bayesian inference. Cameron J. Fackler (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, facklc@alum.rpi.edu), Alistair Hurrell (Dept. of Mech. Eng., Univ. of Sheffield, Sheffield, United Kingdom), Douglas Beaton, and Ning Xiang (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

In noise control and other applications, porous sound-absorbing materials may be constructed of multiple layers of homogeneous materials. This work applies Bayesian inference to study and analyze such multilayer porous materials. The analysis utilizes a measurement of the overall material's acoustic surface impedance and a transfer-matrix porous material acoustic model. The number of layers present in the multilayer porous absorber under test and the physical properties of each layer are inversely determined. Bayesian model selection implements Occam's razor to determine the number of layers present in the sample, based on the measured impedance data and without any a priori knowledge of the number of layers. Once the number of layers has been determined, Bayesian parameter estimation inversely determines the physical properties of all layers simultaneously. A Markov chain Monte Carlo method, nested sampling, is applied to explore efficiently the high-dimensional parameter space inherent to this inverse problem. This sampling method automatically implements the model selection and parameter estimation components to analyze practical multilayer porous absorbers.

3:50

2pNS7. Dose-effect relationships for annoyance due to road traffic noise: Multi-level regression and consideration of noise sensitivity. Laure-Anne Gille (Direction territoriale Ile de France, Cerema, 21-23 rue Miollis, Paris Cedex 15 75732, France, laure-anne.gille@cerema.fr) and Catherine Marquis-Favre (Univ Lyon, ENTPE, Laboratoire Génie Civil et Bâtiment, Vaulx-en-Velin, France)

An *in situ* survey was performed in eight French cities in 2012 to study the annoyance due to combined transportation noise sources. The European Union dose-effect relationships were compared to these new survey data for noise annoyance due to road traffic noise. The measured annoyance was not satisfactorily predicted by these curves: only the percentages of people highly annoyed by road traffic noise was well predicted. Following a multi-level regression as used to construct the European Union dose-effect relationships, new dose-effect relationships were proposed. These new dose-effect relationships enabled a better calculation of noise annoyance due to road traffic noise. Finally, a methodology to consider noise sensitivity in the computation and the percentage of people sensitive to noise in the results is proposed, as this non-acoustical factor is well known to influence noise annoyance. However, the results showed that taking into account such variable did not enable to enhance the dose-effect relationships.

4:10

2pNS8. Uncertainty estimates of psychoacoustic thresholds obtained from group test. Jonathan Rathsam and Andrew Christian (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, jonathan.rathsam@nasa.gov)

A common goal in laboratory psychoacoustic testing is determining the level, or threshold, at which a test signal is judged subjectively equivalent to a reference signal. Adaptive methods, in which the next signal level depends on the response to the previous signal, are most efficient and precise for determining thresholds but only accommodate one subject at a time. At NASA, testing one subject at a time to achieve a sample size representative of a larger population would exhaust testing resources. Instead, three or four subjects are tested simultaneously using preselected signal levels. When a threshold is estimated from a curve fit through group test data, techniques are required for assessing the threshold uncertainty. In this presentation, we examine the Delta Method, the Generalized Linear Model (GLM), the Nonparametric Bootstrap, and Bayesian Posterior Estimation (BPE). Each technique is first exercised on a manufactured, theoretical dataset. When we are confident we are using the methods correctly, we apply them to two psychoacoustic datasets. The Delta Method is the simplest to implement. The BPE is the most versatile, allowing the inclusion of prior information. While useful, the Nonparametric Bootstrap takes longer to calculate. The GLM is found to be least robust.

4:30–4:50 Panel Discussion

TUESDAY AFTERNOON, 24 MAY 2016 SALON H, 1:00 P.M. TO 4:10 P.M.

Session 2pPA

Physical Acoustics and Biomedical Acoustics: Vortex Beams and Radiation Torque Physics II

Philip L. Marston, Cochair
Physics and Astronomy Dept., Washington State University, Pullman, WA 99164-2814

Likun Zhang, Cochair
University of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199

Invited Papers

1:00

2pPA1. Numerical calculation of acoustic radiation force and torque on non-spherical particles in Bessel beams. Felix B. Wijaya and Kian Meng Lim (Mech. Eng., National Univ. of Singapore, Dynam. Lab E1 02-03 National University of Singapore 1 Eng. Dr. 2, Singapore 117576, Singapore, a0107285@u.nus.edu)

Manipulation of biological cells using acoustic radiation force has drawn a lot of attention in recent years. The force and torque acting on cells are usually estimated from analytical and semi-analytical solutions derived for simple shapes, such as spheres and ellipsoids, typically in an axisymmetric configuration. Since biological cells come in various shapes and sizes and they may have an arbitrary orientation in a microfluidic channel, there is a need for a more versatile and robust numerical model for evaluating the acoustic radiation force and torque. Motivated by this, a three-dimensional boundary element model is developed for calculating radiation force and torque on particles of arbitrary shapes and sizes subjected to arbitrary acoustic waves. The first order acoustic field is solved by using the boundary element method. The second-order, time-averaged tractions are then obtained from the first order field. Subsequently, the resultant radiation force and torque are calculated by integrating the tractions over the surface of a fictitious sphere that encapsulates the particle. The force and torque on non-spherical particles subjected to acoustic Bessel beams are obtained using this numerical model. The effects of the beam cone angle and particle orientation on the radiation force and torque are investigated.

1:25

2pPA2. Acoustic forces acting on particles and fluids in microscale acoustofluidics. Henrik Bruus and Jonas T. Karlsen (Dept. of Phys., Tech. Univ. of Denmark, DTU Phys., Bldg. 309, Kongens Lyngby 2800, Denmark, bruus@fysik.dtu.dk)

Acoustofluidics relying on acoustic forces to handle fluids and particles in microfluidic systems has emerged as a useful tool for characterizing, focusing, separating, and sorting cells based on their acousto-mechanical properties. Here, we present recent advances in the theoretical understanding of acoustic forces on particles and fluids. In particular, we address the effects of thermoviscous boundary layers on acoustic scattering off sub-micron particles or droplets. Re-examining the far-field method of calculating acoustic radiation forces and torques, we show that exact non-perturbative expressions can be derived regardless of boundary layer dissipation. The necessary condition for moving the surface of integration from the particle surface to the far field, is the time-periodicity of the system rather than negligible boundary-layer dissipation. In the long-wavelength limit, this approach leads to particularly simple expressions for the force and torque acting on a particle in a thermoviscous fluid. Finally, relaxing the requirement of having two immiscible phases (particle/fluid or droplet/fluid), we generalize the theory to include acoustic forces acting on continuous density and compressibility distributions of inhomogeneous fluids, such as aqueous salt solutions.

1:50

2pPA3. Singular acoustics: Transfer of orbital angular momentum to matter. Régis Wunenburger (Univ. of Paris 6, Paris, France) and Etienne Brasselet (Univ. of Bordeaux, LOMA (UMR5798), 351 cours de la libération, Talence 33400, France, etienne.brasselet@u-bordeaux.fr)

A general feature of wave physics is the existence of phase singularities. One usually refers to beams carrying phase singularities as vortex beams, which have become more popular in the field of optics than in acoustics. In particular, the first experimental demonstration of transfer of orbital angular momentum of sound to matter came much later than its optical counterpart. Here, we will review our recent results regarding the transfer of acoustic orbital angular momentum to matter in the ultrasonic domain. This includes (i) the quantitative test of acoustic orbital angular momentum transfer to a sound absorbing object, (ii) the introduction of a novel phenomenon named "rotational acoustic streaming," and (iii) the demonstration a nondissipative sound-matter orbital angular momentum transfer mediated by chiral scattering.

2:15

2pPA4. On-chip generation of acoustical vortices with swirling surface acoustic waves for single particle manipulation and vorticity control. Antoine Riaud, Michael Baudoin (IEMN, Université de Lille 1, IEMN, Ave. Poincaré, Villeneuve d'Ascq 59652, France, michael.baudoin@univ-lille1.fr), Jean-Louis Thomas (INSP, Sorbonne universités, Univ. Paris 6, Paris, France), and Olivier Bou Matar (IEMN, EC Lille, Villeneuve d'Ascq, France)

Surface acoustic waves (SAWs) are versatile tools for the manipulation of fluids at small scales. These waves can be used to displace, divide, merge, and atomize sessile droplets, but also actuate fluids embedded in microchannels. In this presentation, we will show that IDTs array and inverse filter technique enable on-chip synthesis of a new type of SAWs, called swirling SAWs, which degenerate into bulk acoustical vortices when transmitted to a liquid. These acoustical vortices can be tailored to create a 3D particle trap and thus selective single particle acoustical tweezers, with digital control of the trap position. They can also induce controlled vortical flows, whose topology essentially relies on the topology of the underlying acoustical vortex.

2:40–2:55 Break

2:55

2pPA5. Acoustic radiation and viscous torque for micromanipulation controlled rotation of particles in fluid cavities. Andreas Lamprecht, Thomas Schwarz, Jingtao Wang, and Jurg Dual (Dept. of Mech. and Process Eng., ETH Zurich, Tannenstrasse 3, CLA H23.1, Zurich, Zurich 8092, Switzerland, lamprecht@imes.mavt.ethz.ch)

Our investigations at ETH Zurich aimed at the theoretical analysis of the acoustic torque and its experimental realization of a controlled rotation of spherical and non-spherical particles by ultrasound. Ultrasonic manipulation of particles provides a contactless handling method for particles suspended in a fluid by acoustic streaming and radiation forces. In addition to the translation of particles in all three spatial directions, particles like functional beads, cells, clumps of cells, fibers, etc., can be rotated. Various methods for the rotation of non-spherical particles were developed with the acoustic radiation torque. The necessary varying pressure field, where the orientation of the nodal pressure lines was controlled by two orthogonal standing waves, was achieved by a modulation of one single parameter over time, e.g., amplitude, phase, and frequency. Stable particle and fiber rotations up to 40 rpm were reached. The rotation can be performed continuously or in a stepwise fashion. Moreover, the rotation of spherical objects was realized by the viscous torque. This torque is formed by acoustic streaming, due to two orthogonal standing waves shifted in phase at the same excitation frequency and amplitude. Angular rotations up to 1200 rpm for spherical 35.5 μm copolymer particles were reached.

Contributed Papers

3:20

2pPA6. Acoustic propagation from a helicoidal wavefront source in an ocean environment. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

A helicoidal wavefront source generates a pressure field with a phase dislocation along its axis. About this dislocation, the phase varies linearly with angle while at the dislocation there is a null in the pressure field. The simplest source which can produce this field consists of two crossed dipoles driven 90 degrees out of phase with one another. This phased dipole source was previously used to model propagation from a spiral wavefront beacon in an ocean environment (Hefner and Dzikowicz, JASA, 2012). For the spiral wavefront beacon, the dislocation axis is aligned vertically and the phase variation near the x-y plane is used for underwater navigation. This phased dipole source was shown to be related to the point source through a simple transform which made it possible to transform any point source solution in an ocean environment into the solution for a spiral source in the same environment. This transformation is generalized such that the dislocation axis can be oriented in any direction. This makes it possible to examine helicoidal wavefront propagation in an ocean environment as well. Applications of this general transformation are presented for a helicoidal source near the sea surface and in an ocean waveguide.

3:35

2pPA7. Radiation force expansions in terms of partial wave phase shifts for scattering: Examples. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu) and Likun Zhang (Phys. Dept. & Ctr. for Nonlinear Dynam., Univ. of Texas at Austin, Austin, TX)

When evaluating radiation forces on spheres in acoustic beams, with or without helicity, the interpretation of analytical results is greatly simplified by retaining the use of s-function notation for partial-wave coefficients imported into acoustics from quantum scattering theory in the 1970s. This facilitates easy interpretation of various efficiency factors [L. Zhang and P. L. Marston, Phys. Rev. E. **84**, 035601 (2011); L. Zhang and P. L. Marston, Bio. Opt. Express **4**, 1610–1617 (2013); (E) **4**, 2988 (2013)]. For situations in which dissipation is negligible each partial wave s-function becomes characterized by a single parameter: the partial wave phase shift that parameterizes possible degrees of freedom [P. Marston and L. Zhang, J. Acoust. Soc. Am. **131**, 3534 (2012); P. L. Marston, J. Quant. Spectrosc. Radiat. Transf. **162**, 8–17 (2015)]. The s-function and associated phase shifts are associated with scattering by plane traveling waves, and the incident wavefield of interest is separately parameterized. (When considering allowed outcomes, the method of fabricating symmetric objects having a desirable set of plane-wave scattering partial-wave phase shifts becomes a separate issue.) The present analysis illustrates the advantages of the formulation by extending some prior expansions associated with radiation forces. [Supported in part by ONR.]

3:50–4:10 Panel Discussion

TUESDAY AFTERNOON, 24 MAY 2016 — SALON D, 1:00 P.M. TO 3:55 P.M.

Session 2pPPa

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

Anna C. Diedesch, Cochair
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Adrian KC Lee, Cochair
University of Washington, Box 357988, University of Washington, Seattle, WA 98195

Chair's Introduction—1:00

Invited Papers

1:05

2pPPa1. Differences in cortical responses to spatial changes in foreground versus background auditory objects. Darrin K. Reed, Brigitta Tóth, and Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, dkreed@bu.edu)

Identifying neural markers related to the integration of auditory features—such as spectrum, temporal coherence, and lateral position—can help to reveal the mechanisms underlying how the auditory system processes simultaneous sound sources. For a stimulus that is composed of randomly chosen successive inharmonic tone complexes, a subset of tones can perceptually segregate from the other simultaneous tones if the subset repeats for a sufficient number of tokens. Listeners can reliably detect this repeating pattern as a figure sound object amidst a randomly changing background. An electroencephalogram was recorded while listeners performed the detection of changes in interaural time difference (ITD) cues that were attributed to either the figure or the background. Detection of ITD changes for either auditory object elicited a fronto-central early negativity (100–200 ms) and P300 event-related potentials (ERP) associated with identification of a behaviorally relevant event. However, the ERP amplitudes were larger for the ITD changes associated with the figure compared those associated with an ITD change of the background. This illustrates that the manner in which spectro-temporal features combine to form auditory objects can modulate cortical responses to spatial information. The result supports the notion that spatial features integrate subsequent to object formation.

1:30

2pPPa2. Shining a light on the neural signature of effortful listening. Ian M. Wiggins, Pramudi Wijayasiri, and Douglas Hartley (National Inst. for Health Res. Nottingham Hearing Biomedical Res. Unit, Div. of Clinical Neurosci., School of Medicine, Univ. of Nottingham, Ropewalk House, Nottingham NG1 5DU, United Kingdom, ian.wiggins@nottingham.ac.uk)

This study used functional near-infrared spectroscopy (fNIRS) to investigate a possible neural correlate of effortful listening. Hearing-impaired individuals report more listening effort than their normally hearing peers, with negative consequences in daily life (e.g., increased absenteeism from work). Listening effort may reflect neuro-cognitive processing underlying the recovery of meaning from a degraded auditory signal. Indeed, evidence from functional magnetic resonance imaging (fMRI) suggests that the processing of degraded speech is associated with increased activation in cortical areas outside the main auditory regions within the temporal lobe. However, acoustic scanner noise presents a serious methodological challenge in fMRI. This challenge can potentially be overcome using fNIRS, a silent, non-invasive brain-imaging technique based on optical measurements. In the current study, we used fNIRS to confirm the finding that attentive listening to degraded (noise-vocoded) speech leads to increased activation in the left inferior frontal gyrus (LIFG), compared to a clear-speech control condition. In contrast, robust activation in the auditory cortices was unaffected by speech clarity or attentional focus. Our results support the suggestion that the LIFG plays a role in compensating for degradation to the speech signal. Furthermore, fNIRS may hold promise as a flexible tool to examine the neural processes underlying effortful listening.

1:55

2pPPa3. Temporary unilateral hearing loss during development impairs behavioral and neural sensitivity to interaural level difference cues for sound localization. Kelsey L. Anbuhl (Neurosci. Training Program, Univ. of Colorado Anschutz Medical Campus., RC1-N, Rm. 7401G, 12800 East 19th Ave., Aurora, CO 80045, kelsey.anbuhl@ucdenver.edu), Nathaniel T. Greene, Andrew D. Brown, Victor Benichoux (Physiol. and Biophys., Univ. of Colorado Anschutz Medical Campus., Aurora, CO), Alexander T. Ferber (Neurosci. Training Program, Univ. of Colorado Anschutz Medical Campus., Aurora, CO), and Daniel J. Tollin (Physiol. and Biophys., Univ. of Colorado Anschutz Medical Campus., Aurora, CO)

Children who experience persistent conductive hearing loss (CHL) early in life often display binaural hearing impairments that persist long after CHL is resolved, suggesting abnormal central auditory development. Abnormal sensitivity to interaural level differences (ILDs) is particularly likely as a CHL (such as an ear infection) can attenuate sound in the affected ear by >30 dB, dramatically distorting ILD cues. Here, we quantified the effects of unilateral CHL on (1) behavioral spatial acuity and (2) neural information processing of ILD cues in the guinea pig auditory midbrain (inferior colliculus, IC) using the mathematical framework of Fisher information (FI). Animals raised with unilateral CHL displayed larger minimum audible angles for high-pass noise compared to age-matched controls, suggesting impaired ILD sensitivity. Based on acoustic directional transfer function measurements, ILD discrimination thresholds were elevated by ~3–6 dB. Following behavior, extracellular recordings were made in the IC contralateral to the previously occluded ear, and ILD discrimination thresholds for single neurons were determined using FI. Across the population, neural ILD discrimination was moderately impaired (~2–3 dB worse-than-control) in CHL animals. Impaired processing of ILD in the IC may in part explain the spatial discrimination deficits observed in animals and children with developmental CHL.

2:20–2:40 Break

2:40

2pPPa4. Removing effects of ear-canal acoustics from measurements of otoacoustic emissions. Karolina Charaziak and Christopher Shera (MEEI\HMS, 243 Charles St., Boston, MA 02114, KarolinaCharaziak2013@u.northwestern.edu)

Otoacoustic emissions (OAEs) are the acoustic fingerprints of the inner ear—when carefully measured in healthy ears, their spectra, although highly individualized, remain stable over time. Thus, OAE changes usually indicate changes in cochlear function, e.g., due to efferent modulation, aging, noise trauma, and/or exposure to harmful agents. In humans, however, the reproducibility of OAE measurements is compromised by ear-canal standing waves at relevant frequencies. We show that even when stimulus levels are tightly controlled using methods designed to avoid standing-wave problems (forward-pressure-level calibration), distortion-product (DP)OAE levels vary by ~10–15 dB near half-wave resonant frequencies, depending on probe insertion depth (deep versus shallow). We propose a method, derived from a tube model of the ear canal, that separates the initial outgoing OAE pressure wave at the eardrum from reflected OAEs trapped in the residual ear-canal space. The emitted pressure level (EPL) represents the load-independent OAE level that would be recorded in an ideal, reflectionless canal. When DPOAE levels are converted to EPL, variability across insertion depths decreases by ~10 dB near half-wave resonant frequencies. EPL may provide a simple way to reduce confounding OAE variability across subjects and to improve the reliability of OAE measurements for detecting cochlear changes.

3:05

2pPPa5. Modeling dynamic properties of spontaneous otoacoustic emissions: Low-frequency biasing and entrainment. Dario Vignali, Stephen J. Elliott, and Ben Lineton (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, dv206@soton.ac.uk)

Spontaneous otoacoustic emissions (SOAE) are sounds generated inside the living cochlea and are regarded as by-products of the active mechanism present in the peripheral auditory system. There is still debate whether these emissions are the product of local oscillators located at various positions along the cochlea (local oscillator model) or if they result from standing waves due to a global collective phenomenon that involves different aspects of the cochlea (global standing wave model). This paper uses a global standing wave cochlear model to predict various features of SOAEs. This involves a state-space formulation with a spatially distributed set of nonlinear active micromechanical elements coupled via cochlear fluid coupling. Simulation results have been compared with available experimental data and demonstrate two interesting nonlinear features of the cochlea: first, nonlinear properties of SOAEs modulated by external low-frequency bias tones are easily predicted and can be used to investigate the plausibility of different nonlinear functions incorporated in the micromechanical elements. Second, entrainment patterns can be obtained when the model is stimulated by a swept-tone: results

show distinct areas of beating and others of entrainment between the external stimulus and the SOAE, which depend on the level and instantaneous frequency of the sweep.

3:30

2pPPa6. Hidden hearing loss in tinnitus with normal hearing thresholds. Brandon T. Paul (Psych., Neurosci., & Behaviour, McMaster Univ., 1280 Main St. West, PC 333, Hamilton, ON L8S 4K1, Canada, paulbt@mcmaster.ca), Ian Bruce (Elec. and Comput. Eng., McMaster Univ., Hamilton, ON, Canada), and Larry Roberts (Psych., Neurosci., & Behaviour, McMaster Univ., Hamilton, ON, Canada)

Tinnitus—the phantom ringing of the ears—is thought arise from neuroplastic changes in the central auditory system in response to peripheral hearing damage. However, a minority of tinnitus sufferers have clinically normal hearing thresholds. One explanation of these cases is that auditory nerve fibers with low firing thresholds (LT-ANFs) are intact, but ANFs with high thresholds (HT-ANFs) are not. HT-ANF damage would be "hidden" to the audiogram but evident in suprathreshold tests. To test this hypothesis, we measured the ability of tinnitus and control subjects with normal audiograms to detect amplitude modulation (AM) in a 5 kHz, suprathreshold tone in a narrowband noise. We also recorded by 32-channel EEG the "envelope following response" (EFR, generated subcortically) to the same AM tone in conditions of noise and no noise. Tinnitus subjects had worse AM detection thresholds and had smaller EFRs compared to controls. Simulations of ANF responses from the model of Zilany *et al.* (2014) found that in addition to ~100% loss of HT ANFs, a further ~30% loss of LT fibers was needed to account for the reduced EFRs of tinnitus subjects. ~30% of LT ANFs would not have been expected to affect hearing thresholds. [Work supported by NSERC of Canada.]

TUESDAY AFTERNOON, 24 MAY 2016 SALON H, 4:25 P.M. TO 5:30 P.M.

Session 2pPPb

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Robert P. Carlyon, Chair
CBU, MRC, MRC CBU, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom

Chair's Introduction—4:25

Invited Paper

4:30

2pPPb1. Bridging the chasm: Animal physiology and human psychophysics. Alan Palmer (MRC Inst. of Hearing Res., Univ. Park, Nottingham NG7 2RD, United Kingdom, alan@ihr.mrc.ac.uk)

A goal of animal auditory neuroscience is to understand how humans process and perceive sound. Encouragingly, hypotheses about brain function, generated from psychoacoustics, are sometimes confirmed by physiological experiments. A notable example is the Jeffress model of processing of interaural time differences. This has been largely supported by physiology and indeed we showed that neural activity in the midbrain and cortex, in response to Binaural Masking Level Difference stimuli, was entirely consistent with this cross-correlation model of binaural processing. However, such comparisons of physiology and psychophysics are often beset by caveats of species and anesthesia. This is one reason why the sharpness of human frequency tuning is still currently debated. In recent experiments, we have measured tuning behaviorally and by direct and indirect physiological methods, all in the same species. These different measures are in good agreement, supporting the notion that the cochlea determines perceptual frequency selectivity, and indirectly that perhaps frequency selectivity is different in humans. That notwithstanding, it seems clear that comparisons of animal physiology and human psychophysical measures provide important confirmations and insights into human auditory function.

Session 2pSAa

Structural Acoustics and Vibration: Real-World Instructive Case Studies in Structural Acoustics and Vibration

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

Elizabeth A. Magliula, Cochair
Division Newport, Naval Undersea Warfare Center, 1176 Howell Street, Bldg. 1302, Newport, RI 02841

Chair's Introduction—1:00

Invited Papers

1:05

2pSAa1. What would MacGyver do? Unconventional vibration testing for Mercy Sioux City helistop. Jon W. Mooney (Acoust. & Vib., KJWW Eng. Consultants, 623 26th Ave., Rock Island, IL 61201, mooneyjw@kjww.com)

A trip to Harbor Freight, a hardware store and some MacGyvering—these were essential to arriving at a vibration analysis for a hospital wanting to add a helistop to its roof. Normally not a challenging assignment, the analysis for this project was complicated by many factors, including: the existence of several functioning cath labs on the floor below the roof; unknown vibration and EMI sensitivity of the cath lab equipment; a limited budget— and only one week to prepare for the study. This presentation will explain in detail the goals and constraints of the project, and how three engineers—one an acoustician—developed relatively-simple-yet-ingenious, and low-cost, solutions.

1:25

2pSAa2. Practical aspects of design, tuning, and application of dynamic vibration absorbers. Vyacheslav M. Ryaboy (Newport Corp., 1791 Deere Ave., Irvine, CA 92606, vyacheslav.ryaboy@newport.com)

Theory of dynamic vibration absorbers is a classic topic that has been thoroughly researched. Nevertheless, implementing the dynamic system in a structure and applying it properly still meet some challenges that have not been adequately addressed in the technical literature. Some of practical problems include achieving thermal stability, building a damped device with the desired undamped frequency, providing mechanisms for tuning or adjusting the dynamic absorber *in situ*, and effective tuning of multiple absorbers. The paper describes a new patented design of dynamic vibration absorber for optical tables together with the new method of tuning. The tuning procedure uses only one sensor and does not require access to the moving mass. Applications include suppressing flexural resonance vibrations of standard optical tables and the recent project on vibration control of large support bench for unique petawatt laser system at Lawrence Berkeley National Laboratory.

1:45

2pSAa3. Far field prediction of sound pressure level using panel contribution analysis and scale modeling. David W. Herrin and Gong Cheng (Dept. of Mech. Eng., Univ. of Kentucky, 151 Ralph G. Anderson Bldg., Lexington, KY 40506-0503, dherrin@engr.uky.edu)

Panel contribution analysis is reviewed and then utilized to predict the sound radiation from a generator set. The generator set is divided into patches and transfer functions between the sound pressure at a point in the far field and the velocity of a patch were determined reciprocally both for the full-scale structure and a half-scale model. With the generator set running, a P-U probe was used to measure particle velocity and sound intensity simultaneously on each patch. The sound pressure level at a receiver point located 4.9 m away from the side of the generator set was calculated assuming uncorrelated and correlated sources. The predicted sound pressure, based on both full and half scale models, compared well with that measured. Transfer functions were also determined using boundary element methods and compared well with measurement. In addition, the contributions from each surface of the generator set enclosure were determined.

2:05

2pSAa4. The problem of the noisy golf club. Peter A. Kerrian and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu)

Player perception of the quality of a hand-held sports implement is strongly influenced by the sound generated at impact with a ball. A recent innovative golf club driver design increased the moment-of-inertia of the club, making it more forgiving to hits off center and allowing players to hit balls with greater accuracy. However, this club has not been readily accepted because players hate the loud and annoying sound it makes when striking a golf ball. In this paper, we will discuss the acoustic and vibrational properties of this club as measured through experimental modal analysis, laser Doppler vibrometry, and acoustic field tests. Vibration and acoustic data correlate well and reveal that the annoying high amplitude components in the impact sound are due to vibrational modes in the sole of the club head, at frequencies below that of the "trampoline mode" of the club face. Implications resulting from this study will influence future golf club designs to make them sound more pleasing without altering performance.

Contributed Papers

2:25

2pSAa5. Sound characteristic analysis of sound-pipe-breakwater on the east-coast in Gangneung South Korea. Hyungwoo Park, Won-Hee Lee, and Myung-Sook Kim (SoongSil Univ., 1212 Hyungham Eng. Bldg. 369 Snagdo-Ro, Dongjak-Gu, Seoul, Seoul 156743, South Korea, pphw@ssu.ac.kr)

There is mysterious sounds which can hear after 2010 the new breakwater construction, in a small port city "Yung-Jin" of Gangneung East Sea of Korea. In particular, the mysterious sounds are spread loud honk, when north-east wind are blown in fall and winter of the seasons. People, who have heard the sound that is regarded as a sinister sound, called that like "Kelpie sound," "Sea whistle," and "Pipe sound." This is an ordinary concrete breakwater structure, which has prevented the waves by rocks and tetrapods. The waves are usually considered wind-driven waves. The wind and waves were blocked in rocks and tetrapods. In this time, the wind causes a vortex by the tetrapods. The waterway of the inner harbor and the ocean filter a particular sound out. We can hear and analyze the Mysterious Sound of sound-pipe-breakwater, although the design of the breakwater is not to produce sound. In this study, we investigate and analyze the truth of the strange noises of breakwaters.

2:40

2pSAa6. Ultrasonic imaging with wave mode adaptive weights and global matched coefficient. Simone Sternini, Thompson V. Nguyen, and Francesco Lanza di Scalea (Structural Eng., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, ssternin@eng.ucsd.edu)

This paper discusses some improvements to ultrasonic synthetic imaging in solids. Specifically, the study proposes new adaptive weights applied to the beamforming array that are based on the physics of the propagating waves, specifically the displacement structure of the propagating longitudinal (L) mode and shear (S) mode that are naturally coexisting in a solid. The wave mode structures can be combined with the wave geometrical spreading to better filter the array and improve its focusing ability compared to static array weights. The paper also proposes compounding, or summing, images obtained from the different wave modes to further improve the array gain without increasing its physical aperture. The wave mode compounding can be performed either incoherently or coherently, in analogy with compounding multiple frequencies or multiple excitations. Furthermore, the introduction of a global matched coefficient computed through the matching of measured and expected times of flight will be presented to show additional improvements in the image reconstruction process. Numerical simulations and experimental testing demonstrate the potential improvements obtainable by the wave structure adaptive weights and the global matched coefficient compared to either static weights in conventional delay-and-sum focusing, or adaptive weights based on geometrical spreading alone in minimum-variance distortionless response focusing.

2:55

2pSAa7. Preliminary study of extraterrestrial optical vibrometry for an impactor mission to Europa. Thomas Campbell, John Brewer, Masataka Okutsu, Diego Turo, and Joseph Vignola (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave. NE., Washington, DC 20064, 96campbellt@cua.edu)

We present a methodology to determine the thickness and material properties of the icy crust of Europa, a satellite of Jupiter. Instruments proposed for NASA's Europa Multiple Flyby mission may be insufficient to adequately measure the ice's thickness. We propose a complementary experiment in which vibration of the Europan surface is induced by an artificial impact. Optical vibrometry techniques, such as 2-D digital image correlation and vibration magnification are considered for detecting the displacement field at a sub-pixel level. We make use of COMSOL Multiphysics, which simulates wave motions via the finite element method. Dispersion curves of elastic waves and the wavenumber filtering are used to estimate the thickness and properties of the extraterrestrial ice.

3:10

2pSAa8. A study on a sound fire extinguisher using special sound lens. Iksoo Ahn, Hyungwoo Park, Seonggeon Bae, and Myungjin Bae (TeleCommun. & Information, Soongsil Univ., 369 Sangdo-ro, Dongjak-gu, Seoul 156-743, South Korea, aisbestman@naver.com)

Sound fire extinguisher is developed based on principle of quenching fire by blocking inflow of oxygen with contacting vibration energy from low frequency sound under 100 Hz to fire and then, lowering its temperature. The first sound fire extinguisher made by a Federal Defence researcher was very huge and based on theory of initial principles, making it hard to be commercialized. Although next sound fire extinguisher developed by students of George Mason University was smaller than before, it was also at the basic stage with early theory. What is more, disturbance from the too long cable and inconvenience of having to carry all the tools were not effective to be commercialized. By improving these drawbacks, Sori Sound Engineering Research Institute (SSERI) of Soongsil University has developed new concept sound fire distinguisher. It concentrates sound through special sound lens and increases energy 10 times more to put out the fire. This study researches about reducing weight of sound fire extinguisher based on new concept special sound lens created by Sori Sound Engineering Research Institute of Soongsil University.

Session 2pSAb

Structural Acoustics and Vibration and Signal Processing in Acoustics: Wavenumber Transform Methods

Micah R. Shepherd, Chair

Applied Research Lab, Penn State University, PO Box 30, Mailstop 3220B, State College, PA 16801

Chair's Introduction—3:40

Invited Papers

3:45

2pSAb1. Wavenumber analysis of vibrating dimpled beams. Kyle R. Myers (Appl. Res. Lab., Penn State Univ., 3220B G Thomas Water Tunl, PO Box 30, State College, PA 16804, krm25@arl.psu.edu) and Koorosh Naghshineh (Mech. Eng. Dept., Western Michigan Univ., Kalamazoo, MI)

A structure's noise and vibration characteristics can be changed by introducing dimples onto the surface. These structural modifications are an attractive form of noise control since they are simple to manufacture and do not add mass to the structure. In order to gain a better understanding of the effect of dimples, beams with different number of dimples are considered in this study. The natural frequencies and mode shapes of beams with any number of dimples are computed using a boundary value model derived from Hamilton's Principle. The results of this model are compared to the finite element method and to previous literature. Then, the effect of the dimples on the radiation properties of beams is investigated by examination of their wavenumber spectra. By shifting the wavenumber content of the beam between supersonic and subsonic regions, dimples are able to change the amount of radiation emitted. For some boundary conditions, the wavenumber spectrum is time-dependent as the beam vibrates through a complete cycle. Several examples are presented that demonstrate how different numbers of dimples, as well as their locations and geometries, affect the wavenumber spectra of beams.

4:05

2pSAb2. Achievable performance of a novel Bragg-scattered acoustic receiving array. Benjamin Cray and Ivars Kirsteins (NUWC, 1176 Howell St., Newport, RI 02841-1708, benjamin.cray@navy.mil)

The merits of a novel beamforming technique, based on generating Bragg-scattered acoustic wavelengths along the surface of an array, will be described. The technique, denoted acoustic trace wavelength enhancement, relies on embedding periodic structures within an array, chosen to precisely replicate and shift an incident acoustic wavenumber into higher wavenumber regions. Thus, shorter trace wavelengths are created over the aperture surface. The enhancement technique is documented in two recent publications: enhanced directivity with array grating [J. Acoust. Soc. Am. **136**, 2014] and experimental verification of acoustic trace wavelength enhancement [J. Acoust. Soc. Am. **138**, 2015). These references dealt, however, solely with high signal-to-noise ratios. Specifically, we will investigate the noise characteristics of this new array by calculating its array gain and Cramer-Rao lower bounds on bearing estimation error for plane wave signals embedded in an isotropic Gaussian noise field. Of particular interest is how the performance of the enhanced array compares to a conventional hydrophone-based array.

4:25

2pSAb3. Wavenumber domain analysis of plates with embedded acoustic black holes. Philip Feurtado and Steve Conlon (Appl. Res. Lab., The Penn State Univ. PO Box 30, University Park, PA 16804, paf932@arl.psu.edu)

Acoustic black holes (ABHs) have been developed and demonstrated as effective, passive, lightweight bending wave absorbers that reduce the structural vibration and radiated sound power of plates. By introducing a gradual change in the local plate thickness, the bending wave speed is reduced and the transverse vibration amplitude is increased. Energy can then be effectively dissipated through material losses or attached damping treatments. In this paper, wavenumber spectra were generated from the vibrational responses of a uniform plate, an undamped ABH plate, and a damped ABH plate. The results showed that wavenumber transform analysis is a useful method for investigating and characterizing ABH performance and behavior. The results also demonstrated that ABHs can distribute supersonic bending waves into subsonic wavenumbers. The results will be useful for the design, characterization, and optimization of ABH systems for real structures

4:45

2pSAb4. Processing of defective ring data by "super-interpolators". Rudolph Martinez (Appl. Physical Sci., 49 Waltham St., Lexington, MA 02421, rmartinez@aphysci.com)

We develop an analytical-numerical scheme to make up for full or partial sensor failures in a ring array. The technique's goal is to restore missing information to a recorded signal in the form of a Hermitian covariance matrix for an arbitrary dependent variable. Our analysis breaks the problem down into two: first, by considering the special case of an imperfect array in which the positions and degrees of failure of the flawed sensors are known, and, second, by generalizing that approach to one in which neither piece of information is presumed nor in fact is the knowledge of how many sensors are affected required. An important application of the first part is the retrieval of the perfect signal at equispaced points that have been left un-instrumented for reasons of economy or lack of physical access (a missing sensor becomes equivalent to one with a known null gain). The problem's more general second part has led to a fundamental relationship for a periodic array's total number of sensors, the bandwidth of the ideal signal being restored, and the rank of an integral equation developed from one of Fourier series' dual statements of orthogonality. We call the products of our two-tier technique "super-interpolators" even though neither engages in that activity mathematically.

TUESDAY AFTERNOON, 24 MAY 2016 SALON J, 2:10 P.M. TO 5:25 P.M.

Session 2pSC

Speech Communication, Psychological and Physiological Acoustics, Architectural Acoustics, and ASA Committee on Standards: Intelligibility Challenges: Speakers, Listeners, and Situations

Amy T. Neel, Cochair
Dept. Speech and Hearing Sci., Univ. of New Mexico, MSC01 1195, Albuquerque, NM 87131

Suzanne E. Boyce, Cochair
Department of Comm. Sciences and Disorders, University of Cincinnati, Mail Location 379 University of Cincinnati, Cincinnati, OH 45267

Chair's Introduction—2:10

Invited Papers

2:15

2pSC1. The role of rhythm perception in recognition and learning of disordered speech. Stephanie A. Borrie (Dept. of Communicative Disord. and Deaf Education, Utah State Univ., 1000 Old Main Hill, Logan, UT 84322, stephanie.borrie@usu.edu) and Kaitlin L. Lansford (School of Commun. Sci. and Disord., Florida State Univ., Tallahassee, FL)

The musical advantage describes the idea that musicians, relative to non-musicians, are more successful at deciphering speech in challenging listening conditions. One assumption is that fine-tuned rhythm perception skills in the musical domain translate to fine-tuned processing of rhythm cues in the speech domain. But what happens when the rhythm cues afforded by the speech signal are, themselves, disordered? This study investigated whether the ability to perceive musical rhythms provides a perceptual advantage for recognition (initial intelligibility) and learning (intelligibility improvement) of dysarthric speech—a neurologically degraded acoustic signal characterized by rhythm abnormalities. Fifty young, normal hearing adults participated in two key tests including a rhythm perception test and a perception and learning test with standard pretest, training, and post-test phases. Initial intelligibility scores for each participant reflected words correct on the pretest, and intelligibility improvement scores reflected post-test words correct minus pretest words correct. The results revealed that rhythm perception scores predicted intelligibility improvement scores but not initial intelligibility scores. Findings are discussed in relation to theoretical models regarding the link between music and speech processing, and offer direction for new models that consider the perceptual consequence of rhythm abnormalities in disordered speech.

2:35

2pSC2. Speech intelligibility for native and non-native talkers and listeners. Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198, rajka@mail.utexas.edu)

Much of daily communication occurs under a range of suboptimal conditions related to talker, listener, and signal characteristics. These "adverse" conditions (Mattys *et al.*, 2012) increase intelligibility variation and impact perceptual processes, representations, attention, and memory functions. In this talk, I focus on two factors that condition intelligibility variation, namely, when interlocutors are non-native speakers of a target language and when communication occurs in noise. First, I will discuss a set of experiments examining how native and non-native talkers modify their speech with the goal of enhancing their intelligibility and the extent to which these modifications enhance word recognition in noise for native and non-native listeners. I also present results from studies examining how listeners adapt to more and less familiar foreign accents of varying intelligibility when presented in noise. These studies provide insights into the interplay between physical and mental factors in the production and perception of intelligible speech. They further reveal constraints on talker-listener adaptation processes during communication. A detailed understanding of speech intelligibility variation for native and non-native talkers and listeners will enhance our understanding of the compensatory and cognitive mechanisms that allow speech comprehension in naturalistic conditions.

2:55

2pSC3. Listener profiles for dysarthric and accented speech. Amy T. Neel (Dept. Speech and Hearing Sci., Univ. of New Mexico, MSC01 1195, Albuquerque, NM 87131, atneel@unm.edu)

In order to determine which aspects of speech contribute most to speech intelligibility, listeners rated several speech features for sentences produced by speakers with dysarthria and for passages read aloud by speakers of English as a second language. For the dysarthric speech, speech components included rate, stress, intonation, articulation, voice quality, and nasality in addition to speech intelligibility. For the accented English passages, the speech components were rate, stress, intonation, articulation, and other aspects of speech as well as overall intelligibility, accentedness, and language competence. For both types of speech, articulation was the best predictor of intelligibility ratings. Voice quality, rate, and nasality made minor contributions to intelligibility ratings for dysarthric speech. Rate also contributed slightly to intelligibility ratings for the accented English passages. Potential differences in the salience of speech components for listeners will be discussed. Use of the perceptual profile approach for planning treatment and assessing treatment efficacy for dysarthric and accented speech will also be addressed.

3:15

2pSC4. Applications of speech intelligibility assessment. Karen Payton (Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747-2300, kpayton@umassd.edu)

There are many situations when it would be helpful to be able to predict how intelligible a talker will be to a specific listener in a specific environment. Metrics such as the Speech Intelligibility Index (SII) [e.g., Ricketts, Ear Hear. **17**, 124–132 (1996)] and the Speech Transmission Index (STI) [e.g., Humes *et al*., J. Speech Hear. Res. **29**, 447–462 (1986)] have been used for such purposes. This talk will review the history, philosophy, and effectiveness of these and other metrics in specific situations affecting the chain between talker and listener. Some of the situations include: hearing loss [e.g., HASPI: Kates and Arehart, Speech Commun. **65**, 75–93 (2014)], degradations due to room acoustics [e.g., SRMR: Santos *et al*., Speech Commun. **15**, 815–824 (2013)], and assistive communications technologies. The role of assumptions regarding speech in normal versus disorder populations will be briefly touched on.

3:35–3:50 Break

3:50

2pSC5. Industry testing of voice quality output from mobile devices. Carol Y. Espy-Wilson, Xinhui Zhou, and Ali Akrouf (OmniSpeech LLC, A.V. Williams Bldg., College Park, MD 20742, espy@umd.edu)

In general, the commercial evaluation of speech for telecommunication is focused on what is called “speech quality” rather than intelligibility per se. This talk will discuss the current state-of-the-art in the context of the objective test model 3QUEST (3-fold Quality Evaluation of Speech in Telecommunications) developed by Head Acoustics. 3Quest is an objective evaluation of transmitted speech with background noise and we will discuss the new ETSI standard that it is based on. The particular improvement relative to previous objective methods (PESQ and POLQUA) is that influence of different background noises is taken into account and the calculation of three mean opinion score (MOS) values, one each for the speech-only regions of the signal (SMOS), the noise-only regions of the signal (NMOS), and the total signal (GMOS). These scores allow for a more meaningful statement regarding the causes for the impression of quality. Using examples, we will discuss how signal noise, intelligibility, and quality are interrelated as signals are affected by various parameters.

4:10

2pSC6. Auditory and cognitive aging: Differences between speech intelligibility and speech-in-speech intelligibility. Margaret K. Pichora-Fuller (Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, k.pichora.fuller@utoronto.ca)

Older adults, even those with normal or near-normal audiograms, report greater subjective listening difficulty in everyday life compared to younger adults. However, their difficulties are not well predicted from pure-tone or speech audiometry in quiet or noise. Speech intelligibility tests based on word recognition do not incorporate many of the sensory and cognitive challenges that listeners confront in everyday situations where it is necessary to understand a target talker when there are competing talkers in the auditory scene. Speech-in-speech intelligibility is affected by age-related differences in auditory and cognitive processing. Age-related differences in auditory temporal processing reduce access to periodicity and temporal envelope cues that serve stream segregation and spatial listening. Age-related differences in cognitive processing are reflected in difficulty remembering speech heard in multi-talker babble or switching spatial attention when there is uncertainty about the location of a target talker in a multi-talker display. Furthermore, when listening occurs in multitasking conditions (e.g., listening while walking), there are increased and competing demands for cognitive resources that may affect performance on listening and/or the competing tasks. These examples highlight the importance of both inter- and intra-individual differences in speech-in-speech intelligibility that depend on the interaction of auditory and cognitive processing abilities.

4:30

2pSC7. Production and perception of clear speech. Ann Bradlow (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, abradlow@northwestern.edu)

When talkers are aware of listeners' speech perception difficulties due to hearing loss, noise, or a language barrier, they typically adopt an intelligibility-enhancing speaking style known as "clear speech." To the extent that clear speech is more intelligible than "conversational speech," a cross-style acoustic-phonetic comparison provides information about factors that affect speech intelligibility. I will present data from a project that tested the hypothesis that clear speech is guided by both universal, auditory-perceptual factors and language-specific, structural factors. The former serve to enhance the overall acoustic salience of the speech signal such that it is more resistant to the adverse effects of noise or listener-related perceptual deficits; the latter serves to enhance the realization of phonological contrasts. This hypothesis predicts that clear speech production shows predictable and systematic cross-language similarities and differences, and that the clear speech intelligibility benefit is modulated by the listeners' experience with the target language. To address these predictions, we present data from a cross-language comparison of clear speech production and a cross-population comparison of clear speech perception. These studies provide fundamental information about variability in speech intelligibility which may ultimately lead to effective, efficient, and listener-specific speech intelligibility enhancement strategies.

4:50

2pSC8. Lexical and nonlexical approaches to intelligibility in speech: An intermediate approach. Suzanne E. Boyce (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Mail Location 379 University of Cincinnati, Cincinnati, OH 45267, Suzanne.Boyce@uc.edu)

Intelligibility is a continuum with a floor and a ceiling. In this session, we are concerned with different ways of defining and quantifying intelligibility for different acoustic applications. A major break point in the continuum is defined by lexical transcription. For most applications, such as evaluating clinical and non-native speaker populations, the focus is on whether a reasonably adept listener would come up with a correct transcription under ideal conditions. For other applications, as in telecommunications or clear speaking style, we consider the effects of less-than-ideal conditions, cognitive fatigue, or listener pleasure. In general, however, the quantification of intelligibility is either defined by listener estimates, listener transcription or by broad statistical trends. In this paper, we consider an intermediate method of measuring the likely intelligibility of speech using acoustic characteristics specific to syllable shape, but not tied to transcription. Examples from measuring speaker fatigue, clear speaking style, and disordered source characteristics will be provided.

5:10–5:25 Panel Discussion

TUESDAY AFTERNOON, 24 MAY 2016 SOLITUDE, 1:00 P.M. TO 5:10 P.M.

Session 2pSP

Signal Processing in Acoustics: Comparison of Beamforming, Matched-Field Processing, and Time Reversal Techniques

Brian E. Anderson, Chair
N145 Esc, Brigham Young Univ., MS D446, Provo, UT 84602

Invited Papers

1:00

2pSP1. An overview of beamforming, matched-field processing, and time reversal techniques. William A. Kuperman (Scripps Inst. of Oceanogr., Univ. of California, San Diego, Marine Physical Lab., La Jolla, CA 92093-0238, wkuperman@ucsd.edu)

Beamforming (BF), matched-field processing (MFP), and time reversal (TR) are intimately related. On the simplest level, MFP is a generalization of BF in which the replicas are solutions to a wave equation descriptive of the environmental propagation physics as opposed to BF in which the replicas are typically plane waves. The TR process, on the other hand, is usually taken to be an active process, *independent of computer modeling,* in which the received signals at the elements of an array are time reversed and *physically* retransmitted. In the literature, one often finds examples for which the process of retransmission is implemented by a computer simulation. This is simply a linear version of BF or MFP. Further, passive TR does not refer to this latter case but is a special case. In all cases, there are adaptive processing augmentations. An interesting take on TR is that when TR works, it can be thought of as a kind of proof of existence that the medium supports the localization goal of BF and/or MFP. We review the connection between these various processors.

1:20

2pSP2. Out-of-band beamforming and matched field processing. David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109-2133, drd@umich.edu), Brian M. Worthmann (Appl. Phys., Univ. of Michigan, Ann Arbor, MI), and Shima H. Abadi (Mech. Eng., Univ. of Washington – Bothell, Bothell, WA)

Beamforming and matched field processing (MFP) are array signal-processing techniques for estimating wave propagation directions and source locations, respectively. Both techniques can be formulated as spatial filtering operations that weight and combine the array recordings. In conventional plane- and spherical-wave beamforming, the weights are determined from analytical field models for the corresponding wave type. In conventional MFP, the weights are commonly determined from a computational field model that accounts for known propagation complications (reflection, refraction, scattering, diffraction, etc.) in the acoustic environment. Conventional MFP reduces to conventional beamforming when the environment is simple enough to be described by free-space propagation. Even at high signal-to-noise ratios, both techniques have limitations set by the recorded frequencies, the spacing of the array elements, the geometrical extent of the array, and mismatch between the recorded and modeled acoustic fields. Interestingly, these limitations can be overcome through signal processing techniques that recover out-of-the-signal band information from in-the-signal-band recordings. Here, in-band and out-of-band beamforming and MFP results are illustrated and compared using propagation simulations in free-space and multipath environments, and array recordings from a laboratory water tank and ocean propagation experiments. [Sponsored by the Office of Naval Research and the National Science Foundation.]

1:40

2pSP3. Particle-like wave packets in complex scattering systems. Benoît Gérardin, Jérôme Laurent (ESPCI ParisTech, PSL Res. Univ., CNRS, Université Paris Diderot, Sorbonne Paris Cité, Institut Langevin, 1 rue Jussieu, Paris 75005, France, benoit.gerardin@espci.fr), Philipp Ambichl (Inst. for Theor. Phys., Vienna Univ. of Technol. (TU Wien), Vienna, France), Claire Prada (ESPCI ParisTech, PSL Res. Univ., CNRS, Université Paris Diderot, Sorbonne Paris Cité, Institut Langevin, Paris, France), Stefan Rotter (Inst. for Theor. Phys., Vienna Univ. of Technol. (TU Wien), Vienna, Austria), and Alexandre Aubry (ESPCI ParisTech, PSL Res. Univ., CNRS, Université Paris Diderot, Sorbonne Paris Cité, Institut Langevin, Paris, France)

Waves propagating in complex media typically undergo diffraction and multiple scattering at all the inhomogeneities they encounter. As a consequence, a wave packet suffers from strong temporal and spatial dispersion while propagating through a scattering medium. This wave scattering is often seen as a nightmare in wave physics whether it be for focusing, imaging, or communication purposes. Controlling wave propagation through complex systems is thus of fundamental interest in many areas, ranging from optics or acoustics to medical imaging or telecommunications. Here, we study the propagation of elastic waves in a cavity and a disordered waveguide by means of laser interferometry. We demonstrate how the direct experimental access to the information stored in the scattering matrix of these systems allows us to selectively excite stationary scattering states and wave packets that follow particle-like bouncing patterns in transmission through (or in reflection from) a complex scattering landscape. Due to their limited dispersion, these particle-like scattering states will be crucially relevant for all applications involving selective wave focusing and efficient information transfer through complex media.

2:00

2pSP4. Multiple snapshot and multiple frequency compressive matched field processing. Kay L. Gemba, William S. Hodgkiss, and Peter Gerstoft (MPL/SIO, UCSD, Univ. of California, San Diego, 8820 Shellback Way, Spiess Hall, Rm. 446, La Jolla, CA 92037, gemba@ucsd.edu)

Matched field processing is a generalized beamforming method which matches received array data to a dictionary of replica vectors to locate and track a source. Its solution set generally is sparse since there are considerably fewer sources than replicas. The problem is also underdetermined since the number of sensors are less than the number of unique depth-range cells. Using compressive sensing (CS), the traditional spatial matched-filter problem is reformulated as a convex optimization problem subject to a row-sparsity constraint (RSC). The RSC selects the best match among the replica dictionary when using multiple snapshots. It is found that CS performance is equivalent to the Bartlett processor when comparing the sparse solution to the ambiguity surface's peak for any number of snapshots in a single frequency—single source scenario. Results also indicate that CS performs similarly to the adaptive white noise constraint processor in a multiple source scenario. The RSC can further be exploited to select a common depth-range cell from snapshots corresponding to multiple frequencies to improve the source tracking in the presence of data-replica mismatch. Results are demonstrated using both simulated and SWellEx-96 experiment data.

2:20

2pSP5. Localization of aeroacoustic sources by using time-reversal and beamforming techniques. David Marx, Vincent Valeau, Christian Prax, and Laurent-Emmanuel Brizzi (Institut PPRIME, UPR3346 Bat B17 Campus sud 6 rue marcel dorée, Poitiers 86022, France, laurent.brizzi@univ-poitiers.fr)

The study deals with the localization of aeroacoustic sources in flows by using array processing techniques. In such situations, the source is located in the flow and the propagation model must properly take into account flow effects, with possible temperature gradients. The resolution of the inverse problem can rely: (i) either on the numerical simulation of the propagation by using the time-reversal principle; (ii) or on using the beamforming technique, provided a high-frequency model of the Green function is used, such as that given by a simplified analytical model or by ray-tracing. The advantages and links between the two approaches are discussed. Some comparisons in terms of localization error are then provided, based on simulated data, both in the time and frequency domains. The case of a pulse in a shear flow is considered, showing similar performances for both methods. Then a harmonic source in a shear flow and in a jet flow is studied, with possible temperature gradients, showing the limitations of the beamforming method compared to the time-reversal-based method. Finally, examples of applications to experimental or numerical data are discussed.

2:40–2:55 Break

2:55

2pSP6. Acoustic beamfolding: New potentials enabled by interfacing reconfigurable origami and acoustic structures. Danielle T. Lynd and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., E540 Scott Lab, Columbus, OH 43210, lynd.47@osu.edu)

Acoustic transducer arrays yield large spatial and spectral change in energy transmission through strategic positioning of planar array elements. Yet, unless array elements are actively tuned through phase delay controls, the array performance characteristics remain fixed and suitable for limited purposes. Origami, the art of paper folding, is a means to introduce enormous topological change through simple, kinematic translation and rotation of connected, planar facets, which is one reason for its growing attention as a fluent vehicle to remarkably adapt system properties for multifunctional purposes. In this research, we integrate acoustic transducer array development and origami design principles to establish a new framework for adaptive acoustic energy shaping effected by simple geometric and kinematic folding relations. From a flat-folded strip to an unfolded plane, our theoretical and experimental results show that this transducer design concept leads to powerful means to tune the significance of radiated acoustic energy across orders of magnitude and to easily tailor the acoustic power transmission to the far field.

3:10

2pSP7. The effect of transducer directivity on properties of time reversal focusing. Miles Clemens, Matthew L. Willardson, and Brian E. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, theexperiment113@gmail.com)

This presentation explores the effect of transducer directivity on the focusing properties of time reversal acoustics. Time reversal is a signal processing technique that can be used to focus sound to a location in space for source reconstruction, communication, and intentional sound focusing. One loudspeaker and one microphone are used to produce a time reversal focus in a reverberation chamber. The primary axis of the loudspeaker is oriented in various different directions to produce a time reversal focus for each orientation. Since source directivity depends on the frequency of the sound emitted, these experiments are conducted for a low frequency band and a high frequency band such that these two frequency bands have fairly different directivities. Properties of these foci are compared to determine the overall effect of directivity on the time reversal focus.

3:25

2pSP8. Use of time reversal focusing to create a high amplitude focus of sound in air. Matthew L. Willardson, Miles Clemens, and Brian E. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, mwillardson@verizon.net)

Time reversal is a technique that may be used to achieve intentional sound focusing. The purpose of this presentation is to explore how intense the focus of airborne sound energy can be using time reversal. Experiments utilize loudspeakers and a single microphone placed in a reverberation chamber. Various frequency ranges and loudspeaker orientations are explored along with different types of loudspeakers in order to experimentally determine the optimum conditions for generating the loudest focus possible. The overall purpose of this study is to create an acoustic focus at nonlinear sound pressure levels in order to create a controllable source of nonlinear acoustic sound.

3:40

2pSP9. Comparison of inverse techniques for reproducing an extended, partially coherent sound field above a reflecting plane. Kevin M. Leete, Tracianne B. Neilsen, Blaine M. Harker, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, KML@byu.edu)

The relative abilities of two advanced array processing techniques to reconstruct a partially correlated noise source similar to a high-speed jet are compared. M-SONAH is a modification of statistically optimized near-field acoustical holography that includes multiple types of wavefunctions [A. T. Wall *et al.*, J. Acoust. Soc. Am., **137**, 963–975 (2015)]. The hybrid method may be described as an advanced cross-beamforming algorithm with regularization [T. Padois *et al.*, AIAA Paper 2013-2212]. A numerical, multiple wavepacket-based source distribution provides a reasonable model of the frequency-dependent directivity and coherence properties of a jet noise sound. An image source was included to account for a reflecting ground plane and the numerically modeled sound field was used an input to both the M-SONAH and hybrid methods. For M-SONAH, the image source was handled using an additional set of cylindrical wavefunctions and for the hybrid method, sources were assumed to be located both above and below the reflecting plane. While both methods provide accurate reconstructions, for this test case, the hybrid method performs better at higher frequencies using a realistic measurement aperture and density. [Work sponsored by the USAFRL SBIR program.]

3:55

2pSP10. Time reverse imaging of tsunami waveforms. Jakir Hossen, Phil R. Cummins (Res. School of Earth Sci., Australian National Univ., Canberra, ACT, Australia), and Jan Dettmer (Res. School of Earth Sci., Australian National Univ., 3800 Finnerty Rd., Victoria, Br. Columbia V8W 3P6, Canada, jand@uvic.ca)

We consider the application of Time Reverse Imaging (TRI) to tsunami waveform data, in order to recover the initial sea surface displacement associated with the tsunami source. We use as a case study the tsunami triggered by the March 11, 2011, Tohoku-Oki earthquake, for which an unprecedented number of high-quality observations are available. The method represents the tsunami source by dividing the source region into a regular grid of "point" sources. For each of these, a tsunami Green's function (GF) is computed using a basis function for sea surface displacement whose support is concentrated near the grid point. We apply the TRI method to estimate initial sea surface displacement at each source grid point by convolving GFs with time-reversed observed waveforms recorded near the source region. The result for initial sea surface displacement obtained via TRI agrees well with other models obtained using more traditional inversion methods. We show that the TRI method has potential for application in tsunami warning systems, as it is computationally efficient and can be used to estimate the initial source model using near-field data, and this TRI source model provides accurate and realistic predictions of far-field tsunami waveforms.

4:10

2pSP11. "Knocked Over!": A visual demonstration of time reversal focusing using bending waves in a thin plate. Sarah M. Young, Christopher Heaton, and Brian E. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, sarahmyoung24@gmail.com)

The purpose of this research is to develop a visual demonstration of time reversal focusing in a thin plate to be used as a teaching tool. This presentation will discuss the various plate materials tested to provide the optimal conditions for time reversal focusing. Specifically, the reverberation time in each plate and the sensitivity of the source of vibrations are quantified to provide the best spatially confined focus as well as the highest focal amplitude possible. A vibration speaker and a scanning laser Doppler vibrometer (SLDV) are used to provide the time reversal focusing. A series of objects are placed on the plate in an attempt to knock over objects near the focal location and not elsewhere. Spatial mapping of the plate vibration using the SLDV will be shown to illustrate why some objects fall and others do not.

4:25

2pSP12. On the alternative objective functions for minimum variance distortionless response technique in order to restore the original source. Yuling Tzeng and Gee-Pinn J. Too (Dept. of Systems and Naval Mechatronic Eng., National Cheng Kung Univ., No. 1 University Rd. NCKU, Tainan, Taiwan, yulingtzeng@gmail.com)

Minimum Variance Distortionless Response (MVDR) technique is one of the signal processing ways for noise reduction. Its principle is to minimize the output power of the signal. This study simulated an underwater acoustic field using ray acoustic method. The acoustic field consists of a signal source, multiple noise sources and a line array hydrophones. The signal source is ship-radiated noise. The simulated combining signals received by the hydrophone array are used to restore the original signal source by MVDR technique. The study discusses the different objective functions given in MVDR in order to restore the original source. Then, the results are compared in computing time, effectiveness and adaptability. The results indicate that MVDR can be used to restore the original signal source with very high correlation coefficient up to 90% between the original signal and the restored one.

4:40

2pSP13. Application of time reversal mirror and minimum variance distortionless response technique for source localization. Gee-Pinn J. Too and Yuling Tzeng (Dept. of Systems and Naval Mechatronic Eng., National Cheng Kung Univ., No. 1 University Rd. NCKU, Tainan 70101, Taiwan, toojames@yahoo.com)

Time difference of arrival (TDOA) and time reversal mirror (TRM) are commonly used in acoustic source localization. TDOA has excellent results in free field, but fair results in multiple reflections environment. TRM is applicable in multiple reflections environment and for multiple sources localization. The study continues the advantage of TRM and combines MVDR technique of noise reduction in order to find source locations. The results indicate that the combined MVDR technique gives better noise reduction than TRM in multiple reflections environment. From the drawing of energy diagram of acoustic waveguide, the combined MVDR technique will focus energy on signal source and it also reduces the noise energy in the other region. Comparing between TRM and the combined MVDR for source localization, the combined MVDR gives better resolution than TRM does. In addition to give better resolution, the combined MVDR will find the source location by finding the maximum energy location, on the other hand, TRM is unable to reduce reflection interference effectively which might lead to misjudge the source locations. However, the combined MVDR needs more computing time than TRM does.

4:55

2pSP14. Underwater acoustic communication in time-varying channel environment based on passive time reversal. Xiao Han, Jingwei Yin, Ge Yu, and Xiao Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Bldg., Nantong St., Nangang District, Harbin, Harbin 150001, China, hanxiao1322@126.com)

Channels estimated from probe signal or training symbols will mismatch with the true channels in passive time reversal (TR) underwater acoustic (UWA) communication when the ocean environment is time-varying, leading to poor channel equalization performance. In order to solve this problem, this paper studies Block-based TR processing structure. The received data are divided into small blocks and the channel estimated from previously decoded symbols is used to match filter the current block to suppress the channel mismatch. Decision feedback equalizer (DFE) is also combined with Block-based TR to remove the residual inter symbol interference (ISI) and Doppler effect (Block-based TR-DFE). Mobile UWA communication experiment was conducted near Xiao Changshan island, Dalian in July, 2015. Though the received data are preprocessed by resampling, residual Doppler effect still exists due to ununiform motion between the receiver and transmitter, making the channel time-varying fastly. Block-based TR realized effective tracking and compensation for time-varying channels. The output signal-to-noise ratio (SNR) after equalization is greatly improved and the bit error rate (BER) is significantly reduced.

Meeting of Accredited Standards Committee (ASC) S3/SC 1, Animal Bioacoustics

D. S. Houser, Chair ASC S3/SC 1
National Marine Mammal Foundation, 2240 Shelter Island Drive Suite 200, San Diego, CA 92106

K. Fristrup, Vice Chair ASC S12
National Park Service, Natural Sounds Program, 1201 Oakridge Dr., Suite 100, Fort Collins, CO 80525

Accredited Standards Committee S3/SC 1 on Animal Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43/SC 1 Noise and ISO/TC 43/SC 3, Underwater acoustics, take note that these meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 24 May 2016.

Scope of S3/SC 1: Standards, specifications, methods of measurement and test, instrumentation and terminology in the field of psychological and physiological acoustics, including aspects of general acoustics, which pertain to biological safety, tolerance, and comfort of non-human animals, including both risk to individual animals and to the long-term viability of populations. Animals to be covered may potentially include commercially grown food animals; animals harvested for food in the wild; pets; laboratory animals; exotic species in zoos, oceanaria, or aquariums; or free-ranging wild animals.

TUESDAY AFTERNOON, 24 MAY 2016 ALTA, 3:15 P.M. TO 4:30 P.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

C. J. Struck, Chair ASC S3
CJS Labs, 57 States Street, San Francisco, CA 94114 1401

P. B. Nelson, Vice Chair ASC S3
Department of SLHS, University of Minnesota, 115 Shevlin, 164 Pillsbury Drive S.E., Minneapolis, MN 55455

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note-those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 19 May 2015.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance, and comfort.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

R. J. Peppin, Chair ASC S12
5012 Macon Road, Rockville, MD 20852

A. A. Scharine, Vice Chair ASC S12
U.S. Army Research Laboratory, Human Research & Engineering Directorate
ATTN: RDRL-HRG, Building 459, Mulberry Point Road, Aberdeen Proving Ground, MD 21005 5425

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note that these meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 24 May 2016..

TUESDAY EVENING, 24 MAY 2016 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Tuesday are as follows:

Engineering Acoustics (4:30 p.m.)	Salon A
Acoustical Oceanography	Salon J
Animal Bioacoustics	Salon I
Architectural Acoustics	Snowbird/Brighton
Physical Acoustics	Salon H
Psychological and Physiological Acoustics	Salon B/C
Structural Acoustics and Vibration	Salon G