Session 4aAAa

Architectural Acoustics: Architectural Acoustics Potpourri

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Contributed Papers

8:00

4aAAa1. Comparisons between the preferences of musicians and non-musicians in response to varying room acoustics using two different testing methods. Martin S. Lawless and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, msl224@psu.edu)

Although the brains of musicians and non-musicians differ both on an anatomical and functional level, these groups tend to experience similar emotions in response to the same, unaltered musical excerpts (Bigand/Poulin-Charronnat *Cognition* 2006). Contrarily, preliminary results of the authors and colleagues suggest that the two groups have diverse preferences when presented with the same excerpt convolved with different room conditions. A more comprehensive investigation was conducted to determine differences in preference between musicians and non-musicians in response to stimuli of with different reverberation times, ranging from anechoic to very reverberant. Room acoustics models were used to create auralizations for a number of motifs. Subjects initially rated the stimuli in terms of overall preference, followed by rating their perception of reverberance. Two distinct subjective testing methods, successive and comparative, were utilized and compared. The successive method required the participants to rate each stimulus separately in succession, while the comparative method allowed the subjects to compare and rate each stimulus within a set with the rating scale for each stimulus on one screen. The comparison between methods was performed to validate future testing conducted with the same stimuli in more constrained settings, specifically in a functional magnetic resonance imaging (fMRI) scanner.

8:15

4aAAa2. Investigating the effect of arrival time of diffuse reflections on listener envelopment. Brandon Cudequest, M. Torben Pastore, and Jonas Braasch (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, bcudequest@gmail.com)

Listener Envelopment (LEV) is a quality of diffuse sound fields, and a highly sought after attribute of performance venues. However, the ability for an enclosure to achieve ideal, diffuse sound field is often difficult, and varies from space to space. Thus, a rigid transition time between early and late energy is an insufficient way of evaluating LEV. Current theories on listener envelopment focus on the arrival time of the reflections within the impulse response, but typically disregard the diffusivity of these reflections, building on the fact that the impulse response generally becomes more diffuse over time. An alternative model is proposed, where the listener envelopment is determined by both the diffusivity and the arrival time of reflections. The model disregards the current 80-ms criterion and also allows reflections earlier then this to contribute to LEV. A 64-channel wave field synthesis system is used to perceptually evaluate the effects of spatially and temporally diffuse sound components as a function of arrival time.

8:30

4aAAa3. Connecting the sense of envelopment to specific components of the sound field using perceptually motivated auralizations. Matthew T. Neal and Miche C. Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mtn5048@psu.edu)

Envelopment is known to be a key attribute related to overall room impression. Despite this importance, limited research has been done to identify the specific components of the sound field that contribute to envelopment. The goal of this study was to determine the timing and spatial distribution of reflections contributing to envelopment. A subjective study was conducted using a range of simulated auralizations, which were played back over a three-dimensional loudspeaker array in an anechoic chamber. For each auralization, subjects rated their perceived envelopment. A real-time acoustic simulation program was developed in Max to generate the signals, which simulated early sound with the image-source method and late sound with statistical reverberation. When creating the stimuli, the program produced immediate auditory feedback in response to adjusting the input parameters. The signals were quantified through impulse response measurements to ensure a wide range of conditions. The subjects' envelopment ratings were correlated to different components of the sound field, to evaluate how specific arrival time of reflections and spatial characteristics contribute to envelopment. These results could possibly be used to determine the effectiveness of existing envelopment metrics and potentially contribute to developing a new measure to predict envelopment. [Work supported by NSF Award-1302741.]

8:45

4aAAa4. Image source model for small room acoustics. Ambika Bhatta, Charles Thompson, and Kavitha Chandra (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, ambika_bhatta@student.uml.edu)

In this investigation, the impulse response obtained using exact solution comprising of the Inverse Laplace transform (ILT) of the direction cosine of the incident angle is undertaken. The results are compared to those obtained by Allen and Berkley (JASA **65**, 943–950, 1979). In addition, frequency dependent wall impedance effects are considered. Numerical efficiency and accuracy of the image solution are evaluated.

9:00

4aAAa5. Marketing architectural acoustics to non-acousticians (a.k.a. the unwashed masses). Samuel V. Diaquila (Architecture, Kent State Univ., 21863 Aurora Rd., Cleveland, OH 44146, sam@baswaphon.com)

Architects understand aesthetic issues: color, texture, glare, serenity, excitement, as well as mechanical issues; fire suppression, air changes, and restroom requirements. Their thought process rarely considers the intangible world of acoustics. Their clients (the "owner") is even further removed. In

today's construction environment, how does the acoustical consultant market his or her services successfully to decision makers who are faced with ever decreasing budgets and an ever increasing bombardment of information? Even more challenging is ensuring that the acoustical consultant's recommendations are implemented. Learn ways to awaken your potential client's sensitivity and awareness of acoustics. Changing your sales presentation from a technical (acoustical consultants are generally "blind"), relying on computer generated acoustical reports that resemble mathematical thesis papers to an emotion based technique (potential clients are generally "deaf"), that brings the science of acoustics to life at a tangible human level. Everyone reacts to bad acoustics, with the exception of concert halls and restaurants, acoustics is rarely discussed and often not one of the end user's priorities until there is a problem needing a retrofit.

9:15

4aAAa6. The resonance of tapering spiral chambers. Paula Pino (Paulapart, 142 Irving Ave., Apt. 1R, Brooklyn, NY 11237, paul@paulapart.com)

This paper will explore the acoustical resonance of several cochlea-inspired sculptures (i.e., spiraling, tapering forms) made of glass, ceramics, and other materials. Each sculpture will function as an acoustic chamber and will be equipped with a loudspeaker and a "mouth" through which sound will pour out. Using frequency sweeps, feedback loops, reverb convolution, and acoustic prediction software I will measure the resonant and reverberant responses of these acoustic sculptures and compare them with analogous forms (e.g., animal horns, brass instruments, and mammalian cochlea). Special focus will be on the fundamental resonant frequencies and bass response of each shape. Extrapolation with these data will inform the production of larger, more architectural acoustic chambers modeled with similar spiraling geometry.

9:30

4aAAa7. Architectural acoustics illustrated. Michael Ermann (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu)

Novel discoveries, non-obvious findings, and counter-intuitive conclusions will be excerpted and highlighted from the just-released book *Architectural Acoustics Illustrated* (authored by the presenter, Wiley, 2015). The text filters the content of architectural acoustics through the graphic and built language of architecture. In writing the book, building material choices, spatial relationships, best-practices, and data were explored through drawing and animated videos. Summaries of the findings will be demonstrated graphically to establish relationships in sound absorption, room acoustics, sound isolation, and noise control.

9:45–10:00 Break

10:00

4aAAa8. Simulation and auralization of a concert hall's inhomogeneous sound field using finite difference time-domain methods and wave field synthesis. Kelsey Hochgraf, Jonathan Botts, Ning Xiang, and Jonas Braasch (Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, hochgk@rpi.edu)

Auralization methods have been used for a long time to simulate the acoustics of a concert hall for different seat positions. The goal of the research project presented here is to apply the concept of auralization to a larger audience area that the listener can walk through to compare differences in acoustics for a wide range of seat positions. For this purpose, the low frequency acoustics of Rensselaer's Experimental Media and Performing Arts Center (EMPAC) Concert Hall are simulated using finite difference time-domain (FDTD) methods to create signals for a 128-channel wave field synthesis (WFS) system located at Rensselaer's Collaborative Research Augmented Immersive Virtual Environment (CRAIVE) Laboratory. By allowing multiple subjects to dynamically experience the concert hall's acoustics at the same time, this research gains perspective on what is important for achieving objective accuracy and subjective plausibility in a simulation and auralization. Efforts are made to maintain efficiency of wave-based modeling, and methods for evaluating the final auralization are explored from both objective and perceptual standpoints.

10:15

4aAAa9. Practical desktop full-wave architectural acoustic solutions. Patrick Murray, Jeff Cipolla, and Adam Hapij (Appl. Sci., Weidlinger Assoc., 40 Wall St., FL19, New York, NY 10005, patrick.murray@wai.com)

An explicit, time-domain, finite-element method is shown to calculate the reverberation time in realistic acoustic spaces using desktop computing resources. The reverberation time is defined as the time required for reflections of direct sound to decay 60 dB, and is also the principal quantity in architectural acoustics across the frequency ranges of interest. Current industry practice for calculating the reverberation time involves empirically derived formulae which cannot account for the architectural complexity of modern acoustic spaces or the detailed placement of acoustic treatments. Computing advances over the past several years have made it possible to calculate the reverberation time using finite element models in the time domain. Explicit finite-element codes are distinct from traditional ones because they operate by integrating the governing equations of mass, momentum, and energy in the time domain, avoiding the need for matrix generation, storage, and factorization. The feasibility of this full-wave approach is demonstrated using the configurations of real example acoustic spaces. Comparison is made with empirical calculations and experimental measurements. It is also shown how the enhancements can be made to a space by the addition of acoustic damping material.

10:30

4aAAa10. Beyond ISO3382—Measuring acoustics with live sound. David H. Griesinger (Res., David Griesinger Acoust., 221 Mt. Auburn St #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Current acoustic measurements techniques typically require time and heavy equipment. The effort and expense would be justified if the results accurately predicted the sound quality from a performance at an individual seat, but they do not. If one believes that it is possible to hear and accurately report sound quality from live sound in different seats, then it must be possible to measure it this way. This paper describes a model of human hearing that promises to provide this ability, at least for quality aspects relating to clarity. The model is based on the ease with which information is carried from one or more sources to a listener. For speech, this involves both reverberant masking from late reverberation, and interference to the direct sound from early reflections. To make such a measurement, we need to model how the ear and brain system precisely localizes sound sources, and separates their sound from each other and acoustic interference without added information from context, grammar, or prior knowledge. A description of such a model will be presented, along with results obtained from live sound.

Session 4aAAb

Architectural Acoustics and Education in Acoustics: Up and Coming Architectural Acousticians: Past Student Paper Award Winners Report

Lauren M. Ronsse, Cochair
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David T. Bradley, Cochair
Physics + Astronomy, Vassar College, 124 Raymond Avenue, #745, Poughkeepsie, NY 12604

Invited Papers

9:45

4aAAb1. Acoustic adventures over continents, among cultures: Under tides of academia and profession. Zühre Sü Gül (MEZZO Studyo LTD, METU Technopolis KOSGEB-TEKMER No112, ODTU Cankaya, Ankara 06800, Turkey, zuhre@mezzostudyo.com)

After obtaining a Bachelor of Architecture and an MFA with an emphasis in amphitheater acoustics, I was accepted into Rensselaer Polytechnic Institute's unique Graduate Program in Architectural Acoustics. There I completed an MS degree focusing on architectural acoustics in coupled spaces and was honored to receive the Robert Newman Award, a TCAA Student Paper Award and an ASA commendation for student design. Since 2006 I have applied my academic focus to design projects, first at RBDG Inc. in Dallas, and later at home in Turkey at MEZZO Stüdyo, the firm I co-founded in 2009. Turkish governmental supports have afforded MEZZO the opportunity to become a critically recognized "R&D incubator." We've also completed over one hundred design projects, both in-house and teamed with architectural-engineering groups around the world and are proud members of NCAC (National Council of Acoustical Consultants). During the restricted times of not managing my firm—and my 3-year-old son!—I enjoy giving university lectures to inspire future generations about exciting diversities in the field of acoustics. Luckily, I just completed my PhD degree on Architectural Acoustics again, and I am much proud and happy to share this whole adventure.

10:00

4aAAb2. Strategies for guiding undergraduate acoustics students through independent research projects. Lauren M. Ronsse (Audio Arts and Acoust., Columbia College Chicago, 33 E. Congress Pkwy, Ste. 601, Chicago, IL 60605, lronsse@colum.edu)

Students in the B.S. Acoustics Program at Columbia College Chicago must successfully complete an independent research project as part of the curriculum. Since this is the first independent research experience for most of the students, measures are taken to guide them through the research process. Also, an electronic research notebook tool has been developed, providing a structured format for the students to document their research. The teaching techniques employed to prepare and guide the undergraduate acoustics students through the research process will be described. Recent student research projects that utilized the electronic research notebook reporting format will also be highlighted.

10:15

4aAAb3. Cultureshock Los Angeles: A midwesterner's guide to acoustics, food trucks, and surfing. Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, srawlings@veneklasen.com)

Samantha Rawlings & Joshua Magee were awarded a Student Paper Award at the Fall 2007 Acoustical Society of America conference in New Orleans for undergraduate research on sound radiation from footfall noise. This student project paved the way for Samantha to move to Los Angeles to join Veneklasen Associates, which was a wholly un-looked for opportunity for someone who once laughed outright at a Hardees commercial showing a hamburger wrapped in lettuce. Since moving to LA, Samantha has become a LEED AP and Project Manager and has had the chance to work on projects of all size and scope; everything from the "mystery noise" cold call to landmark project such as City Creek Center in Salt Lake City and Wilshire Grand in downtown Los Angeles. Additionally, Samantha has participated in research projects that have been presented at ASA, NoiseCon, and Internoise conferences. In this presentation, Samantha reflects on the surprising events of the past seven years and the adventure of learning to fit in and love Southern California.

10:30

4aAAb4. Parallels in scattering research between architectural and underwater acoustics. Derek R. Olson (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, dro131@psu.edu)

Acoustical scattering is a central area of research in the fields of underwater and architectural acoustics. Although the mathematical formulation of the scattering problem is common to both fields, their goals and environments have motivated divergent analysis techniques, and quantities associated with the pressure field due to scattering. One of the primary differences is the role of statistical averaging.

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Ensemble-averaged quantities can often be used in ocean acoustics because the seafloor may have statistically homogeneous roughness and geoacoustic properties over large areas. In the built environment, the acoustic field can interact many times with a single acoustic diffusor, and deterministic properties of the scattered field are of greater import. Acoustical quantities such as the scattering cross section, scattering coefficient, and the diffusion coefficient will be defined and compared. Examples will be given that highlight cases when ensemble-averaged quantities can aid or hinder design of acoustical diffusors, depending on the design constraints.

10:45

4aAAb5. Effects of conductor verbal and nonverbal instructions on changes in singer voicing behaviors: A survey of recent research. Jeremy N. Manternach (College of Education/School of Music, The Univ. of Iowa, 311 Communications Ctr., Iowa City, IA 52242, jeremy-manternach@uiowa.edu)

Vocal and choral music professionals seek to guide their singers to efficient and beautiful vocal production. They do so using a multitude of techniques, some of which have been passed along from one generation of teachers to the next. These techniques include giving singers a particular focus of attention and displaying varied conducting gestures designed to evoke specific voicing behaviors. Despite the prevalence of these techniques, few researchers have investigated whether they consistently change the acoustical output or the muscle engagement used by individual singers or choristers. This presentation provides an overview of recent research that has sought to quantify the voicing behaviors of individual singers as they (a) sang a short melody while focusing on various internal or external foci, (b) viewed varied conductor preparatory (inhalation) gestures prior to singing a short melody, and (c) viewed varied conductor final release (cutoff) gestures at the end of a short melody. The resulting acoustical, biomedical, and perceptual data inform the pedagogical implications that are discussed throughout.

11:00

4aAAb6. Continuing research endeavor in room acoustic effects on speech communication and psychoacoustics. Z. Ellen Peng (Inst. of Tech. Acoust., RWTH Aachen Univ., RWTH Aachen, Kopernikusstraße 5, Aachen, Germany, ellen.z.peng@gmail.com)

I received the best student paper award in architectural acoustics at the San Francisco ASA meeting in 2013. I presented the first half of my dissertation project on designing acoustics for linguistically diverse classrooms by investigating the effect of room acoustics on speech comprehension by native and non-native English-speaking listeners. In the following year, I continued to complete my dissertation and received my PhD in architectural engineering from the University of Nebraska-Lincoln. Beginning in 2015, I am a postdoctoral fellow at RWTH Aachen University in Germany as part of the iCARE (icareitn.eu) research program funded by a Marie-Curie Initial Training Network (ITN) grant from the European Union. My role is to help improve an existing electronic system in extending its application of virtual acoustics testing on children with hearing impairment. Upon receiving the best student paper award, I continue to enjoy multidisciplinary research that combines architectural acoustics, speech communication, and psychoacoustics.

11:15

4aAAb7. An approach to communicating in-field acoustical performance to architects as it relates to end user experience. Ari M. Lesser and Adam P. Wells (Acoustics, Cerami & Assoc., Inc., 404 Fifth Ave. 8th Fl., New York, NY 10018, alesser@ceramiassociates.com)

A study was conducted within an industry leading architecture firm to benchmark the noise isolation class, background noise level, and reverberation time throughout the firm's spaces. Testing was conducted in accordance with the general guidelines of ASTM E336-14, ANSI S1.13, and ASTM E2235-04. Acoustical test results were compared to employee subjective satisfaction survey responses in an effort to bridge the gap between acoustical design recommendations and end users' expectations and experiences of their environment. [Work supported by Cerami & Associates, Inc.]

11:30

4aAAb8. Navigating and developing acoustics research and curriculum within the Georgia Tech Center for Music Technology. Timothy Hsu (School of Music, Georgia Inst. of Technol., 840 McMillan St., Atlanta, GA 30332-0456, timothy.hsu@music.gatech.edu)

Acoustics at the Georgia Institute of Technology has primarily been based within mechanical engineering with various other faculty spread throughout the university. Within the last decade, the School of Music has developed an internationally recognized graduate degree program and research center in music technology. Since winning the Student Paper Award, my professional career has started in a unique faculty position within the School of Music at Georgia Tech, where my responsibilities are split between being an ensemble/choral conductor and an active member of the Center for Music Technology. One of the recent challenges is the development of an innovative undergraduate curriculum model that fuses music fundamentals, music technology, general musicianship, and engineering/computer science. At the graduate level, new coursework I have created has included a musical acoustics course for music technologists and a historical acoustics and modeling course. Additionally, developing a research track of acoustics within the confines of music technology has allowed my research to morph into musically related areas that my graduate research did not allow. Areas of current research include historical architectural acoustics, active temperament, synthesis, and noise control.

11:45

4aAAb9. Teaching acoustics to people with non-scientific backgrounds. Ana M. Jaramillo (Ahnert Feistel Media Group, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu)

With a background in architecture, the author has spent some time developing curriculum for the teaching of acoustics to architecture, design, and music technology majors. Acoustic software is often employed as a tool for the understanding of acoustic concepts, especially helpful when teaching people with non-scientific backgrounds. Through her work with Anhert Feistel Media Group (AFMG), the author has had a chance to take a closer look at acoustic software and spend some time understanding the needs of acoustic simulation software users and learners.

Session 4aAO

Acoustical Oceanography: General Topics in Acoustical Oceanography

Zoi-Heleni Michalopoulou, Chair
Mathematical Sciences, New Jersey Institute of Technology, 323 ML King Blvd., Newark, NJ 07102

Chair's Introduction—8:30

Contributed Papers

8:35

4aAO1. Fate of methane gas bubbles emitted from the seafloor along the Western Atlantic Margin as observed by active sonar. Liam Pillsbury and Thomas C. Weber (CCOM, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, LPillsbury@ccom.unh.edu)

A method for characterizing and quantifying marine methane gas seeps along the Western Atlantic Margin (U.S. East Coast) was developed and applied to 70 free-gas seeps observed by the R/V Okeanos Explorer in 2012 and 2013, in water depths ranging from 300–2000 m. Acoustic backscatter from an 18 kHz split-beam echo sounder and a 30 kHz multi-beam echo sounder provided information on the height to which the gas seeps rose from the seabed. Profiles of the depth-dependent target strength and scattering strength per unit depth were generated from the acoustic data. These profiles were compared to models of the evolution of rising bubbles in order to help constrain the ultimate fate of the bubbles. Of particular interest are comparisons of profiles for seeps originating below, at, and above the gas hydrate stability zone.

8:50

4aAO2. Diameter and density dependent target strength of submerged oil droplets measured by a broadband, high-frequency echo sounder. Scott Loranger (Earth Sci., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, sloranger@ccom.unh.edu) and Thomas C. Weber (Mech. Eng., Univ. of New Hampshire, Durham, NH)

Over two million tons of oil enters marine environments from anthropogenic sources annually with severe environmental consequences. The most effective method of cleaning spills is to biodegrade them by dispersing the oil as droplets. However, the ultimate fate of dispersed oil in the environment is largely unknown. Acoustic remote sensing may offer a means by which to assess the quantity, characteristics, and ultimate fate of submerged oil droplets. To provide a foundation for this work, we have made a series of laboratory measurements using a broadband, high-frequency, calibrated echo sounder. Measurements of oil droplet frequency-dependent target strength were made in a 6 m deep tank of fresh water. Target strength was measured and compared to droplet size and density. Droplet size ranged from 60 μm to 1 mm and was measured by high definition camera. Oils of different density were used including castor, gasoline, diesel, and crude oil. Sound speed of each oil was measured using a Digibar Pro sound velocimeter.

9:05

4aAO3. Laboratory observations of the target strength of non-spherical gas bubbles in water. Thomas C. Weber, Liam Pillsbury, and Scott Loranger (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, tom.weber@unh.edu)

Naturally occurring methane bubbles in the ocean are often observed to have ellipsoidal or otherwise deformed shapes. Models of acoustic scattering from non-spherical gas bubbles suggest changes in resonance frequency and scattering strength relative to spherical gas bubbles with identical volumes. These changes potentially confound our ability to easily relate measurements of acoustic backscatter from gas bubbles to their size and quantity. To help quantify the magnitude of this effect, we have conducted a series of laboratory measurements of acoustic backscatter from non-spherical air bubbles rising from the bottom of a 6 m deep test tank. The acoustic measurements were made at frequencies between 10 kHz to 150 kHz, well above the frequency of bubble resonance (as is often the case for measurements of methane bubbles in shallow coastal environments). Laboratory measurements of bubbles with different sizes and deformations are compared with models for spherical bubbles with identical volumes.

9:20

4aAO4. Low frequency scattering from dynamic fish schools based on collective animal behavior modeling. Simón E. Alfaro, Jorge Cellio (Ingenieria, Universidad Tecnologica de Chile, Av. Vitacura 10.151, Vitacura, Santiago - Chile, Santiago 7650033, Chile, simon_alfaroj@hotmail.com), Maria P. Raveau (Ingenieria Hidraulica y Ambiental, Pontificia Universidad Catolica de Chile, Santiago, Chile), and Christopher Feuillade (Instituto de Fisica, Pontificia Universidad Catolica de Chile, Santiago, Chile)

Low frequency acoustic scattering from swim bladder fish is dominated by the monopole resonance response of the bladder. A school scattering model has previously been developed [Feuillade *et al*., J. Acoust. Soc. Am. **99**(1), 196–208 (1996)] to predict levels of scattering from schools of bladder fish, which includes multiple scattering effects among the fish, and coherent summation of their radiated fields. In order to incorporate these acoustic interactions, the relative locations of the individual fish within the school are required as an input. To provide a realistic description of fluctuating levels of scattering from schools, a self-organizing model of group formation in three-dimensional space has been developed, based on biological principles of collective animal behavior [Couzin *et al*., J. Theor. Biol. **218**, 1–11 (2002)]. In this model, organization within the school is a function of alignment, and repulsive and attractive tendencies based upon the position and orientation of the individual fish. The results of using this model to simulate the fish behavior demonstrate the spatial and temporal dynamics of the fish school, and indicate how these influence the statistical variability of the acoustic scattering response as a function of frequency. [Work supported by ONR.]

9:35

4aAO5. Bayesian environmental inversion of airgun modal dispersion using a single hydrophone in the Chukchi Sea. Graham A. Warner, Stan E. Dosso, Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Ste. A405, Victoria, British Columbia V8P 5C2, Canada, sdosso@uvic.ca), and David E. Hannay (JASCO Appl. Sci., Victoria, British Columbia, Canada)

This paper presents estimated water-column and seabed parameters and uncertainties for a shallow-water site in the Chukchi Sea, Alaska, from trans-dimensional Bayesian inversion of the dispersion of water-column acoustic modes. Pulse waveforms were recorded at a single ocean-bottom hydrophone from a small, ship-towed airgun array during a seismic survey. A warping dispersion time-frequency analysis is used to extract relative mode arrival times

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as a function of frequency for source-receiver ranges of 3 and 4 km, which are inverted for the water sound-speed profile (SSP) and subbottom geoacoustic properties. The SSP is modeled using an unknown number of sound-speed/depth nodes. The subbottom is modeled using an unknown number of homogeneous layers with unknown thickness, sound speed, and density, overlying a halfspace. A reversible-jump Markov-chain Monte Carlo algorithm samples the model parameterization in terms of the number of water-column nodes and subbottom interfaces that can be resolved by the data. The estimated SSP agrees well with a measured profile, and seafloor sound speed is consistent with an independent headwave arrival-time analysis. Environmental properties are required for anthropogenic noise modeling studies in the Chukchi Sea and for improving acoustic localization of marine mammals detected with passive acoustic monitoring systems.

9:50

4aAO6. Measuring the acoustic scattering response of small groups of live fish in a laboratory tank. Maria P. Raveau, Christopher Feuillade (Pontificia Universidad Católica de Chile, Vicuña Mackenna 4860, Macul, Santiago 7820436, Chile, mpraveau@uc.cl), Gabriel Venegas, and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Assessing the validity of measurement/model comparison in fisheries acoustics is difficult, due to the uncertainty in ground truth for acoustic measurements obtained in the open water. One way to overcome this is to utilize laboratory measurements, where fish school parameters may be more well known. The primary purpose of this work was to investigate the feasibility of measuring the acoustic properties of a small group of live fish in a laboratory tank using a steady state subtraction method [J. Acoust. Soc. Am. **112**, 1366–1376 (2002)]. Transfer function measurements were obtained in a fresh water tank that contained an enclosed group of goldfish (*Carassius auratus auratus*), in order to describe their resonance scattering behavior. The experimental results were compared with an existing predictive model [J. Acoust. Soc. Am. **99**, 196–208 (1996)], which incorporates both multiple scattering effects between fish, and coherent interaction of their individual scattered fields. Computational modeling, experimental details and data/model comparison will be presented. This technique can be extended to larger tanks and other fish species. [Work supported by ONR.]

10:05–10:20 Break

10:20

4aAO7. Application of time-warping to passive acoustic remote sensing. Oleg A. Godin (Physical Sci. Div, CIRES, Univ. of Colorado and NOAA Earth System Res. Lab.,325 Broadway, Mail Code R/PSD99, Boulder, CO 80305-3328, oleg.godin@noaa.gov), Justin S. Ball (CIRES, Univ. of Colorado, Boulder, CO), Michael G. Brown (Rosenstiel School of Marine and Atmospheric Sci., Univ. of Miami, Miami, FL), Nikolay A. Zabotin, Liudmila Y. Zabotina (CIRES, Univ. of Colorado, Boulder, CO), and Xiaoqin Zang (Rosenstiel School of Marine and Atmospheric Sci., Univ. of Miami, Miami, FL)

Interferometry of ambient and shipping noise in the ocean provides a way to estimate physical parameters of the water column and the seafloor without employing any controlled sound sources. With noise interferometry, two-point cross-correlation functions of noise serve as the probing signals and replace the Green's function measured in the active acoustic remote sensing. The amount of the environmental information, which can be obtained with passive remote sensing, and robustness of the estimates of the water column and the seafloor parameters are expected to increase, when contributions of individual normal modes can be resolved in the noise cross-correlation function. Using the data obtained in the 2012 noise-interferometry experiment in the Straits of Florida [M. G. Brown *et al.*, Geophys. Res. Lett. **41**, 5555–5562 (2014)], this paper demonstrates the feasibility of normal mode decomposition of the noise cross-correlation function measured by two hydrophones. The normal modes are resolved by using time-warping, a signal processing technique that has been previously successfully employed to separate normal modes generated by a wide-band compact sound source in shallow-water waveguides. The passively measured dispersion curves of acoustic normal modes are inverted for geoacoustic parameters of the seafloor. [Work supported by NSF and ONR.]

10:35

4aAO8. Spectrum of sound intensity fluctuations due to mode coupling in the presence of moving nonlinear internal waves and bottom parameters estimation. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, Mt. Carmel, Haifa 31905, Israel, bkatsnels@univ.haifa.ac.il), Valery Grigorev (Phys., Voronezh State Univ., Voronezh, Russian Federation), and James Lynch (WHOI, Woods Hole, MA)

Temporal fluctuations of intensity of sound pulses (300±30 Hz) are studied in the presence of nonlinear internal waves (NIW) moving at the some angle with direction of an acoustic track providing mode coupling. In the episode of Shallow Water 2006 experiment, considered in the work, angle between wave front of NIW and source-receiver direction is about 10 degrees. It was shown that there is main maximum in the spectrum measured at fixed depth ~8 cph and set of smaller peaks (~15 cph, ~25 cph, and ~35 cph) in accordance with the theory proposed by authors earlier. Besides, there is maximum in the spectrum of total intensity (summarized over depth or over all hydrophones of vertical line array) at the frequency ~15 cph. In authors opinion, the last one is determined by coupling of propagating modes (excited by the source) having maximal difference in modal attenuation coefficients (in given case modes 2 and 4). Role of horizontal refraction is small. Using the spectrum speed of NIW and bottom attenuation coefficient (in our case ~0.2 dB/wavelength) are determined. Mentioned parameters are in a good agreement with estimations obtained by other methods. [Work was supported by BSF.]

10:50

4aAO9. Direct inversion for sediment sound speed in ocean acoustics. Tao Lin and Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu)

A fast direct approach for estimating sediment sound speed is presented, building on a previously developed technique. The algorithm inverts for the seabed sound speed employing low frequency measurements. The Deift-Trubowitz integral equation is solved by using sound pressure after a Hankel Transform. A modified Born approximation is implemented that improves on previously obtained results, followed by an interpolation method which provides a further improvement. Although the approaches initially rely on assumptions about the sound speed at infinity, we show that the assumptions can be relaxed without affecting the performance. The methods are stable with noise-free data, but problems arise when noise is added to the acoustic field. It is demonstrated that a regularization technique remedies the problem. [Work supported by ONR.]

11:05

4aAO10. Change-point detection for recursive Bayesian geoacoustic inversions. Bien Aik Tan (DSO National Labs., 14, Sci. Park Dr., Singapore, Singapore 118226, Singapore, tbienaik@dso.org.sg), Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA), Caglar Yardim (ElectroSci. Lab., The Ohio State Univ., Columbus, OH), and William S. Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

In order to carry out geoacoustic inversion in low signal-to-noise ratio (SNR) conditions, extended duration observations coupled with source and/or receiver motion may be necessary. As a result, change in the underlying model parameters due to time or space is anticipated. In this paper, an inversion method is proposed for cases when the model parameters change abruptly or slowly. A model parameter change-point detection method is developed to detect the change in the model parameters using the importance samples and corresponding weights that already are available from the recursive Bayesian inversion. If the model parameter change abruptly, a change-point will be detected and the inversion will restart with the pulse measurement after the changepoint. If the model parameters change gradually, the inversion (based on constant model parameters) may proceed until the accumulated model parameter mismatch is significant and triggers the detection of a change-point. These change-point detections form the heuristics for controlling the coherent integration time in recursive Bayesian inversion. The method is demonstrated in simulation with parameters corresponding to the low SNR, 100–900 Hz LFM pulses observed in the Shallow Water 2006 experiment.

Session 4aBA

Biomedical Acoustics: Acoustic Radiation Force in Biomedical Applications II

Mostafa Fatemi, Cochair
Physiology & Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905

Matthew W. Urban, Cochair
Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905

Invited Papers

9:30

4aBA1. Measurement of cardiovascular tissue stiffness with acoustic radiation force. Gregg Trahey (Biomedical Eng., Duke Univ., 136 Hudson Hall, Box 90281, Durham, NC 27708, gregg.trahey@duke.edu)

We have developed qualitative and quantitative methods of imaging the stiffness of myocardial and vascular tissues using acoustic radiation force. In ongoing animal and clinical studies, these methods have been utilized to differentiate vulnerable from stable vascular plaques, to monitor the formation of radio-frequency- and cryo-ablation lesions, and to assess the impact of ischemia, infarct, pre-load, afterload, and coronary artery pressure on cardiac stiffness. We present alternative pulse-sequencing techniques and image reconstruction methods and discuss their impact on image contrast and resolution. We discuss the relevance of these methods in assessing cardiovascular disease and the technical challenges of introducing them into clinical practice.

9:50

4aBA2. New parameters in shear wave elastography. JeanLuc Gennisson, Thomas Deffieux, Mathieu Pernot, Mathias Fink, and Mickael Tanter (ESPCI, CNRS, INSERM, Institut Langevin, 1 Rue Jussieu, Paris 75005, France, jl.gennisson@espci.fr)

In the field of shear wave elastography, a specific technique called Supersonic Shear Imaging (SSI) was developed since almost 15 years. This technique is based on two concepts: By means of the acoustic radiation pressure phenomena shear waves are generated directly within tissues. Then, shear wave propagation is caught in real time by using an ultrafast ultrasound device (up to 20,000 frames/s). As shear wave speed is directly related to stiffness of tissues, such a concept allows to recover elastic maps of organs. Nevertheless, stiffness is not always only sufficient to better understand organs pathologies and behaviors. So, there is a need to add new parameters for a better characterization. In this context, SSI technique can be extended in order to reach new mechanical parameters, which can potentially help physicians. By looking at the shear wave dispersion, viscosity of tissues can be retrieved by using the right rheological model. Elastic anisotropy is recovered by rotating the probe at the surface of the investigated organ. For each position, the shear wave speed is calculated allowing to deduce orientation of fibers. At last, the change in tissue stiffness as a function of the pressure applied over medium, also called acoustoelasticity theory, allows the assessment of the nonlinear elastic properties. The combination of all these new parameters, viscosity, anisotropy, and nonlinearity, with stiffness offer new possibilities of diagnosis for physicians to better understand organs pathologies.

10:10

4aBA3. Using acoustic radiation force to probe tissue mechanical properties: Challenges and strategies. Stephen A. McAleavey (Biomedical Eng., Univ. of Rochester, 309 Goergen BME/Optics Bldg., Rochester, NY 14627, stephen.mcaleavey@rochester.edu)

The acoustic radiation force phenomenon provides a highly flexible instrument with which to probe the mechanical properties of tissue. The ability to shape the applied force both in time and space, combined with the high sensitivity of ultrasound motion tracking, enables a multitude of techniques for the estimation of tissue mechanical properties. A present challenge is to reduce the measurement variance of these properties so as to better distinguish subtle variations associated with disease stage. Shear wave tracking is the starting point for many of these techniques, and can be used to characterize tissue in terms of group velocity, as well in terms of tissue constitutive model parameters. This talk will discuss sources of error in shear wave velocity and tissue parameter estimation and strategies for their avoidance or compensation. Time-domain estimation of viscoelastic parameters using a wave propagation model and maximum likelihood estimator will be described, along with challenges due to non-ideal source geometries.

10:30

4aBA4. Acoustic radiation force to reposition kidney stones in humans. Michael R. Bailey, Bryan W. Cunitz, Barbrina L. Dunmire (Ctr. Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, bailey@apl.washington.edu), Jonathan D. Harper, Franklin H. Lee, Ryan Hsi, Mathew D. Sorensen (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), James E. Lingeman (Dept. of Urology, Indiana Univ. Health, Indianapolis, IN), Maria M. Karzova, Petr V. Yuldashev, Vera A. Khokhlova, and Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

This is a report of the first clinical study to reposition kidney stones with acoustic radiation force. Studies were conducted with the approval of the University of Washington IRB and a U.S. FDA Investigational Device Exemption. Of the 15 subjects, average age was 56 ± 11 years; average BMI was 29 ± 3; and stone size range was dust to 13 mm. Two patients reported skin discomfort and sensation at depth with a few pushes. Otherwise, there was no pain or adverse effects associated with the treatment. Stones were repositioned in 14 subjects. Stones were repositioned to a new location in all 6 post-lithotripsy patients, while 4 of the 6 passed over 30 stone fragments within a few days of treatment. *De novo* stones and stones as large as 8 mm were repositioned. In four of the 15 subjects, what was noted in clinical imaging as a single, potentially unpassable stone was shown to be several passable stones upon repositioning with ultrasound. Ultrasonic propulsion can safely and without pain reposition kidney stones in humans. [Work supported by NIH NIDDK grants DK043881and DK092197 and National Space Biomedical Research Institute through NASA NCC 9-58.]

Contributed Papers

10:50

4aBA5. Simulation of ultrasound radiation force induced shear wave propagation in viscoelastic media using a mapped Chebyshev pseudospectral method. Bo Qiang (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905), John C. Brigham (Dept. of Civil and Environ. Eng., Univ. of Pittsburgh, Pittsburgh, PA), Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI), James F. Greenleaf, and Matthew W. Urban (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN 55905, Urban.Matthew@mayo.edu)

We implemented a three-dimensional solver for simulating the ultrasound radiation force induced shear wave motion in viscoelastic media using a mapped Chebyshev pseudospectral method. The solver used a body force field that is simulated by the Field II program as the excitation. Then, a velocity-stress formulation was used to calculate the shear wave motion. We used a Voigt model to describe tissue viscoelasticity, and the viscosity is introduced by memory variables and auxiliary differential equations. The top boundary of the domain had a fixed boundary condition to mimic the interaction between an ultrasound probe and the tissue surface. All other sides of the domain were surrounded by perfectly matched layers to minimize the reflections. Time stepping was implemented with the LSODE integrator with variable step sizes. Results show that the waveform of the simulated shear wave is close to experimental observations and equivalent FEM simulations. We used a Fourier-based method to estimate the phase velocities of the shear waves between 100–1000 Hz and a nonlinear regression to estimate the Voigt model parameters. The reconstructions matched within 10% error compared with the true values.

11:05

4aBA6. Compensating for Scholte waves in single track location shearwave elasticity imaging. Jonathan Langdon, Karla Mercado, Diane Dalecki, and Stephen McAleavey (Biomedical Eng., Univ. of Rochester, 303 Goergen Hall, Rochester, NY 14627, jonathan_langdon@urmc.rochester.edu)

The estimation of shearwave velocity in biological tissues using Single Track Location Shearwave Elasticity Imaging (STL-SWEI) depends on the assumption that the ultrasonically observed particle displacements are due to the propagation of shearwaves in an approximately infinite space. When this assumption is violated, erroneous estimates of the shearwave speed may occur leading to image artifacts. One particularly troubling error occurs when slowly propagating Scholte waves are generated at solid-fluid interfaces. These interface waves travel at a slower speed than the shearwaves produced in STL-SWEI. However, the signals produced appear similar to that of shearwaves and cannot be readily distinguished in the typical STL-SWEI imaging sequence. Instead, alternative sequences are needed to identify and correct for these anomalous wave types. In this work, the surface wave phenomena is examined in the context of STL-SWEI imaging. The appearance of these waves is demonstrated in simulation, tissue mimicking phantoms, engineered tissues, and in liver tissue using a clinical scanner. The effect of the ultrasound beam geometry on the Scholte wave measurement is studied. Finally, a revised STL-SWEI reconstruction method utilizing Radon Sums is presented. Using this method, the simultaneous characterization of shear and Scholte waves is demonstrated for the above materials.

11:20

4aBA7. Shear wave propagation and elasticity imaging of soft tissues under compression. Dae Woo Park, Man M. Nguyen, and Kang Kim (Dept. of Medicine, Univ. of Pittsburgh, 567 Scaife Hall 3550 Terrace St., Pittsburgh, PA 15213, parkd2@upmc.edu)

In efforts to improve detection sensitivity of shear wave elasticity imaging of target tissue lesions with relatively small mechanical contrast to the background tissues, shear wave propagation characteristics in tissues under compression loading have been studied. A finite element hyperelastic tissue model was constructed to characterize the changes of propagating shear wave subject to different mechanical loading and to guide *in vitro* experiments. The shear wave speed sharply increased in an inclusion from 2.4 m/s to 6.3 m/s while it increased from 2.0 m/s to 4.0 m/s in the background tissue with overall compression loading from 0% to 30%. Increased shear wave reflection at the boundary of the inclusion due to increased mechanical contrast was lowered using a directional filter. *In vitro* experiments were performed using a soft phantom block (0.5% agar with 5% gelatin) that contains a hard inclusion (1.5% agar with 5% gelatin) of a long cylinder (D: 8 mm). The reconstructed shear modulus of the inclusion exhibited noticeable nonlinearity, in contrast to linear increase of shear modulus in the surrounding phantom. As a result, the elastic modulus contrast of the inclusion to the surrounding phantom was increased from 0.47 to 1.41 at compression from 0% to 30%.

11:35

4aBA8. Plane nonlinear shear waves in relaxing media. John Cormack and Mark F. Hamilton (UT Austin, Appl. Res. Lab., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

Due to very low shear moduli for soft tissue or tissue-like media, shear waves propagate very slowly, on the order of meters per second, making it relatively easy to produce shear waves exhibiting waveform distortion and even shock formation. Finite amplitude effects in plane shear waves result from cubic nonlinearity, compared with quadratic nonlinearity in compressional waves. Both attenuation and dispersion also significantly affect propagation of shear waves in tissue. Here we account for these complex viscoelastic effects by considering a medium with one relaxation mechanism. An analytical solution similar to that of Polyakova, Soluyan, and Khokhlov [Sov. Phys. Acoust. **8**, 78 (1962)] for a compressional wave with a step shock in a relaxing medium is obtained for a shear wave with a step shock in a relaxing medium. The wave profile with cubic nonlinearity

closely resembles that with quadratic nonlinearity. For weak nonlinearity the solution reduces to an expression obtained by Crighton [J. Fluid Mech. **173**, 625 (1986)] for a Taylor shock in a viscous medium with cubic nonlinearity. Numerical simulations are presented comparing shock formation with quadratic and cubic nonlinearity for other wave profiles in relaxing media. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

11:50

4aBA9. Three-dimensional finite difference models of shear wave propagation in isotropic, homogeneous soft tissue. Yiqun Yang (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI 48824, yiqunyang.nju@gmail.com), Matthew W. Urban (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN), and Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Shear wave elastography with ultrasound applies an acoustic radiation force to generate shear waves in viscoelastic soft tissue. To address the need for more effective simulation tools that model shear waves generated by an applied acoustic radiation force, three-dimensional (3D) finite difference programs that simulate propagation of shear waves in an isotropic, homogeneous medium have been created. These programs simulate shear wave propagation in elastic and viscoelastic soft tissue models. The 3D finite difference implementation combines an explicit time-stepping approach and a staggered spatial grid with an absorbing boundary condition that reduces boundary reflections. The acoustic radiation force in these simulations is quickly and accurately simulated in FOCUS (http://www.egr.msu.edu/~fultras-web) for a focused linear ultrasound array with an f/# of 2. The compressional wave speed is 1500 m/s and the shear wave speed is 1.5m/s in the elastic and viscoelastic tissue models, and the shear viscosity is 1 Pa·s in the viscoelastic model. The simulation completes 90,000 time steps of a 10 ms simulation in an 80 mm $\times$ 80 mm $\times$ 80 mm region with spatial sampling equal to the wavelength of compressional component in three days on a 3.4 MHz Intel i7 processor. [Supported in part by NIH Grants R01EB012079 and R01DK092255.]

THURSDAY MORNING, 21 MAY 2015 COMMONWEALTH 1, 10:00 A.M. TO 11:05 A.M.

Session 4aEA

Engineering Acoustics and Physical Acoustics: Funding Opportunities with the National Science Foundation

Allan D. Pierce, Chair
PO Box 339, 399 Quaker Meeting House Road, East Sandwich, MA 02537

Chair's Introduction—10:00

Invited Paper

10:05

4aEA1. Sensors, dynamics, and control: Program overview and relevance to acoustics research. Massimo Ruzzene (National Sci. Foundation, 270 Ferst Dr., Atlanta, Georgia 30332, ruzzene@gatech.edu)

The talk will provide an overview of the sensors, dynamics, and control program, its goals, its funding levels, and its priorities. Specifically, the talk will address topics and research directions that are relevant to the acoustics community at large. Such topics include vibration and noise control, dynamic-based structural health monitoring, wave propagation in complex media, nonlinear dynamics, and acoustic metamaterials. Examples of currently funded projects and ideas for future investigations will be provided to facilitate the discussion. The SDC program supports fundamental research on the analysis, measurement, monitoring, and control of dynamic systems, including development of new analytical, computational and experimental tools, and novel applications to engineered and natural systems. Dynamics is the science of systems that change in time. Control concerns the use of external influences to produce desired dynamic behaviors. Diagnostics concerns the use of observation to infer information about a dynamic system. Objectives of the SDC program are the discovery of new phenomena and the investigation of innovative methods and applications in dynamics, control and diagnostics. The intellectual merit of proposals submitted to the SDC program will be evaluated on the basis of fundamental innovation in foundational areas, on alignment with the core disciplines of the CMMI Division, and on potential for transformative impact within and across disciplinary boundaries.

Session 4aED

Education in Acoustics Public Relations, and Student Council: Expanding Acoustics Outreach with Social Media

Andrew A. Piacsek, Cochair
Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926

Andrew C. Morrison, Cochair
Natural Science Department, Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431

Chair's Introduction—8:00

Invited Papers

8:05

4aED1. Twitter's not just for teenagers: A scientists' guide to getting started with Twitter in just 5 minutes a day. Laura Kloepper (Dept. of Neurosci., Brown Univ., 185 Meeting St. Box GL-N, Providence, RI 02912, laurakloepper@gmail.com)

Twitter is an effective, powerful tool for scientists looking to keep abreast of trends in their field, communicate their research, and act as public advocates of science. With a simple 140-character Tweet, a person can announce new publications, comment on current events, share photos and updates of research, connect with budding young scientists, and more. Many people think that using Twitter is a massive time investment, but many basic Twitter activities can be conducted in the time it takes you to drink your morning coffee. This talk will introduce the basics of Twitter, show the power of Twitter, and demonstrate best practices for using Twitter with science—all in just 5 minutes a day.

8:25

4aED2. Understanding the social media medium. Andrew T. Pyzdek (Acoust., The Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu)

With social media platforms like Twitter, Facebook, and Tumblr largely dictating how information is viewed and shared online, Marshall McLuhan's famous statement that "The medium is the message" has never been more true. While it is possible to share the same message over multiple platforms with minimal variation, success is borne when a unique approach is tailored to each one. This talk will focus on the affordances of each social media platform and how those should inform not only your use of social media but also your choice of platform. Special attention will be payed to common uses for social media in acoustics: disseminating original research, establishing connections within the field, advocating science, and educating a general audience on acoustics. Also presented is my personal experience educating a lay-person audience about acoustics as a moderator of the AskScience subreddit and the curator of the Listen To This Noise blog on Tumblr, and how these platforms differ both in content and form.

8:45

4aED3. Using social media tools efficiently and effectively for acoustics outreach and education. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

The ability to quickly and easily reach a large public audience through social media outlets such as Twitter and Facebook makes these platforms ideal for raising awareness of acoustics-related studies and issues. Some researchers may be unable to dedicate a lot of time toward building and maintaining a social media presence. To make finding and disseminating information more efficient on social media sites, there are a series of tools, which can be combined together to automatically aggregate and post information online in ways which can save valuable time and effort. In this presentation, I will show my tool chains that are used to discover interesting acoustics-related news stories for sharing online and how they can be scheduled for posting over several days. I will also discuss surprising ways in which the automatic discovery of acoustics-related questions by non-scientists make for interesting tutorials to be used in acoustics classes at the undergraduate level.

9:05

4aED4. Professional development with social media. Andy Rundquist (Phys., Hamline Univ., 1536 Hewitt Ave., MS B1807, Saint Paul, MN 55104, arundquist@hamline.edu)

I was lonely. I realized that my colleagues didn't always share my particular passions about teaching and research. Reaching out online using twitter, the Global Physics Department, and blogs has helped me find a community that supports me, challenges me, entertains me, and teaches me. Prompts like "Help! I teach normal modes tomorrow and I need better examples" and "how fast do whistlers settle on a frequency" draw me into the community where I hear several, sometimes conflicting, views that then motivate me to learn. My requests about things like whether minimizing the area between a sound wave and a particular sine wave is akin to finding a Fourier coefficient (it isn't) connect me with experts who don't roam my hallways. I'll talk about how I engaged with my various learning communities and how I've developed various workflows to leaverage them.

9:25

4aED5. Using Twitter to disseminate acoustics information: Planning and measurement using marketing science and sociophysics. Lawrence Norris (Arlington Sci. and Technol. Alliance, 6704G Lee Hwy., Arlington, VA 22205, lnorris@arlingtonscience.org)

While many people use Twitter for many different purposes, it is an amazing tool for disseminating scientific information. There are several serious tools for marketing products, information, and people via Twitter. Indeed, a body of science in marketing and sociophysics is available to plan campaigns on Twitter and to measure their efficacy. In this talk, I will discuss tools and methods that the acoustics community can use to engage science students at all levels, their teachers, and parents, as well as other acousticians, policy-makers, and the general public.

9:45

4aED6. @BYUAcoustics: Using social media to enhance research and outreach at BYU. Blaine M. Harker, Tracianne B. Neilsen, Kent L. Gee, Jennifer K. Whiting, Mark L. Berardi, Pauline White, Nicholas D. Ortega, and Matthew F. Calton (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@byu.net)

Social media has become an increasingly common method of attracting the attention of fellow researchers and promoting interest in acoustics worldwide. A concerted effort has been recently made by the Acoustics Research Group at Brigham Young University to promote acoustics with various methods of enhanced communications. Web page articles were developed with lay-language introductions to the group's research activities, which can be shared directly to social media sites. The group's Facebook page and research Twitter account @BYUacoustics provide information about current research meetings, publications, and acoustics in the news, which help network with students who may be interested in joining the acoustics program and which keep alumni informed of current events. In addition, our outreach efforts have been augmented by use of social media. A YouTube channel has been created that contains videos of acoustical demonstration from our outreach show. A general audience Twitter account @Sounds2Astound has been utilized to connect with students in our descriptive class and to K-12 teachers who often bring their students to tour our facilities. The overall effectiveness of each system is assessed using webpage statistics, analytics, and perceived success in reaching target audiences. Successes and limitations are summarized and lessons learned are outlined.

10:05–10:50 Break

10:50

4aED7. The broader impact of practicing communication through social media: From Twitter to the National Science Foundation. Alexis B. Rudd (Alexis Rudd, Univ. of Hawaii at Manoa, 47-420 Hui Iwa St. #B-304, Kaneohe, HI 96158, rudd@hawaii.edu)

Available research funding has decreased in line with the Budget Control Act of 2011, which resulted in funding cuts across both defense and non-defense discretionary programs. These cuts have had a negative impact on many researchers, including those at the Acoustical Society of America. Legislation and appropriations by congress have a direct effect on the funding levels and research priorities of federal agencies. Clear and relatable communication of these impacts is important to both the public and the members of the US Congressional committees, most of whom do not have a background in scientific research. Consideration of the technical and educational background of the audience is vital to clear communication, and agencies such as the National Science Foundation (NSF) are placing increasing emphasis on the broader impacts of scientific proposals and how research will benefit the people of the United States. Social media is an opportunity for scientists to get real-time feedback on science communication and to practice translating scientific jargon for an audience of non-specialists and explaining technical concepts succinctly (often in 140 characters or less).

11:10

4aED8. Reddit Science AMA: Using "the front page of the internet" to share your knowledge with the world! Laura Kloepper (Dept. of Neurosci., Brown Univ., 185 Meeting St. Box GL-N, Providence, RI 02912, laurakloepper@gmail.com) and Andrew Pyzdek (Penn State Univ., University Park, PA)

The social media service and link aggregator reddit is one of the most popular and influential websites on the internet. Boasting 174 million unique visitors per day, a high percentage of those high school and college aged students, reddit provides the perfect audience for science outreach. One of the features of reddit is Science AMA, or Ask Me Anything, in which scientists answer questions about their work. Participation in these Q&As allows scientists to communicate with an international audience of all ages and backgrounds, and is an effective tool for scientific outreach with limited time investment. In this talk, we will share the outcome of a Bioacoustics AMA conducted at the Indianapolis meeting, and learn how more individuals and TCs can participate in future Science AMAs.

11:30–12:00 Panel Discussion

Session 4aNS

Noise, ASA Committee on Standards, Psychological and Physiological Acoustics, Physical Acoustics, and Structural Acoustics and Vibration: Wind Turbine Noise I

Paul D. Schomer, Cochair
Schomer and Associates Inc., 2117 Robert Drive, Champaign, IL 61821

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

Kenneth Kaliski, Cochair
RSG Inc, 55 Railroad Row, White River Junction, VT 05001

Robert D. Hellweg, Cochair
Hellweg Acoustics

Chair's Introduction—8:30

Invited Papers

8:35

4aNS1. Some pitfalls to be avoided in a wind turbine noise research program. Paul D. Schomer (Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

The Acoustical Society of America has created a public policy position relative to the acoustic emissions from wind turbines. This position calls for research that definitively will show if problems exist, and if so, who is affected, how are they affected, and why. Much of the research to date is based on assumptions, frequently contrary to fact or unproven. That is not the kind of research that the ASA desires. The money spent on this questionable research should have been directed towards definitive research such as that envisioned by ASA. This paper talks about some of the previous research and elucidates on their assumptions with the purpose of preventing mistaken test designs like these in the future, and with the purpose of improving the research program to be developed by ASA.

8:55

4aNS2. Health and well-being related to wind turbine noise exposure: Summary of results. David Michaud (Health Canada, Canadian Federal Government, 775 Brookfield Rd., Ottawa, ON J9H7J9, Canada, david.michaud@hc-sc.gc.ca)

This paper summarizes the results of the Health Canada Wind Turbine Noise and Health Study. One participant between the ages of 18–79 years was randomly selected from each household. The final sample included 1238 participates (606 males) living between 0.25 and 11.22 km from wind turbines. The response rate was 78.9% and did not significantly vary across sampling strata or between provinces. Wind turbine noise (WTN) exposure was not found to be related to hair cortisol concentrations, resting blood pressure, resting heart rate, or any of the measured sleep parameters. Self-reported results obtained through an in-person questionnaire do not provide support for an association between increasing WTN levels and self-reported sleep disturbance, use of sleep medication, or diagnosed sleep disorders. Similarly, no significant association was found between WTN levels and self-reported migraines, tinnitus, dizziness, diabetes, asthma, hypertension, perceived stress, or any measure of quality of life. Statistically significant exposure-response relationships were observed between increasing WTN levels and an increase in the prevalence of long term high annoyance towards several wind turbine features, including: noise, shadow-flicker, visual impacts, and vibrations. The influence of background noise on annoyance and the association between WTN annoyance and other reported and measured outcomes is presented.

9:15

4aNS3. Indoor and outdoor narrowband measurements of low frequency and infrasonic noise from wind turbines. Allan Beaudry and Michael Bahtiarian (Noise Control Eng., Inc., 799 Middlesex Turnpike, Billerica, MA 01821, allanb@noise-control.com)

The acoustic impact from industrial-grade wind turbines was studied for turbines located in Cape Cod, Massachusetts. Narrowband and one-third-octave band infrasonic and low frequency measurements were performed at a residence located in proximity to the Vestas V82-1.65 MW wind turbines. Surveys were undertaken inside and outside the residence on multiple occasions under various wind speeds and wind directions. In each case, measurements were performed using an infrasonic microphone prior to, during, and following the nightly shutdown period of the turbines. The spectral results show the existence of discernible tones occurring at the blade passage

frequency and its harmonics during operation and absent with the turbines secured. Additional tones where found in the very low audible frequency range with sidebands separated by the blade passage frequency. Comparisons have been made of the indoor and outdoor measurements as well as the effect of wind speed and direction on tone amplitude and prominence. The use of overall G-weighted noise levels to characterize these low frequency tones is also examined.

9:35

4aNS4. Changes in ambient sound levels observed in relation to the operational status of the Mesa Wind Project Site. Jessica Briggs (Fish, Wildlife, and Conservation Biology, Colorado State Univ., Fort Collins, CO), Dr. Megan F. McKenna, Kurt M. Fristrup, (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CA 80525, kurt_fristrup@nps.gov)

Type 1 sound level measurements and continuous audio recordings were obtained from two sites near the Mesa Wind Project Site (MWPS), 100 m and 1500 m from the nearest turbine. The sites were upwind of the turbines. Unexpectedly, MWPS ceased operating 29 days after this project began; data were collected for an additional 127 days. Contrasts in hourly summaries of 1 second 1/3rd octave sound level measurements, controlled for hourly mean wind speed, revealed the contributions of rotating blades. For the near site, at wind speeds below 5 m/s, 1/3rd octave band levels increased starting at about 100 Hz, with distinct spectral peaks appearing between 400 Hz and 2000 Hz. These spectral peaks were less prominent when hourly mean wind speed was 5 m/s or greater. At the far site, small increases in sound level were observed for 1/3rd octave bands at 1 kHz and below when the blades were turning. The levels measured in this study will be compared with levels measured at NPS sites, to evaluate distances at which operational wind farms might affect park acoustical environments.

9:55

4aNS5. Do wind turbines cause adverse heat effects? A review of the evidence. Robert Y. McMurtry (Surgery, Western Univ., 403 Main St., Picton, OntarioN K0K2T0, Canada, rymcmurtry1@gmail.com)

There is a continuing debate about the veracity of AHE in the environs of WT. Proponents claim there is insufficient evidence to support the existence of direct AHE and therefore WT are safe to erect near human habitation. The international regulations governing WT placement are highly variable but generally a setback of at least 350 m (WHO 500m) and a noise limit of 40 dBA are recommended. Nonetheless there is mounting evidence that AHE are in fact occurring. In the 1980s, N D Kelley produced evidence of direct causation of AHE, especially sleep disturbance, from downwind WT, which was attributable to impulsive infrasound and low frequency noise. Subsequently, there has emerged further evidence regarding upwind WT that satisfies the nine Bradford Hill criteria. The presentation will define AHE and address each Braford Hill criterion.

10:15–10:25 Break

10:25

4aNS6. "Waking in an anxious frightened panicked state" and "pressure bolt sensations"—what clinical and acoustic clues can tell us about the causes of deteriorating physical and mental health. Sarah E. Laurie (Waubra Foundation, PO Box 7112, Banyule, Melbourne, Victoria 3084, Australia, sarah@waubrafoundation.org.au)

Residents living in quiet rural areas report a variety of new symptoms correlating with exposure to new acoustic emissions from industrial sound and vibration sources, including large wind turbines, of which sleep disturbance is the most common. Some residents and their health practitioners report a unique pattern of sleep disturbance, which indicates physiological stress is occurring concurrently. Other clinical and acoustic clues such as "pressure bolt" sensations correlating with independently measured pressure pulses will also be shared. Neurophysiological pathways which appear to be activated by these acoustic stimuli will be discussed, followed by reference to the extensive body of clinical research supporting the current knowledge that both chronic sleep deprivation and chronic stress directly exacerbate pre existing diseases, as well as directly cause new diseases resulting in serious and sometimes irreversible damage to physical and mental health. Finally, specific research to investigate these physiological and pathological impacts, in order to determine safe threshold exposure levels for long term exposure to impulsive infrasound and low frequency noise will be proposed.

10:45

4aNS7. Why are regulators, communities, neighbors, and acousticians annoyed by wind turbines? Stephen E. Ambrose (SE Ambrose & Assoc., Windham, ME) and Robert W. Rand (Rand Acoust., 1085 Tantra Park Circle, Boulder, CO 80305, rrand@randacoustics.com)

Noise assessments for large wind turbines should be more reliable based on decades of human-response research. Empirical data clearly show the most severe noise impacts occur in the quieter environments. Populations living in quiet areas were the least researched until the arrival of wind turbines. European studies reveal that the public has a greater sensitivity to wind-turbine noise than transportation sounds. Acoustic researchers developed the percent highly annoyed metric to model. This metric should be shaped by an assessment of the acoustic environment using many variables all subjective, debatable, and mathematical. A review of dose-response research will attempt to lay the groundwork for a new noise assessment methodology specific to wind turbines.

11:05

4aNS8. Understanding the human impact caused by the sound of wind turbines. William K. Palmer (TRI-LEA-EM, 76 SideRd. 33-34 Saugeen, RR 5, Paisley, Ontario N0G2N0, Canada, trileaem@bmts.com)

A mystery surrounds the human impact reported by those with wind turbines in their environment. "PubMed" identifies 23 papers for "wind turbine human health." A dose-response relationship between exposure to sound, annoyance, and sleep disturbance is generally accepted. Self-reporting identified other impacts that commenced or increased with wind turbines operation. Recent articles repeat that no peer-reviewed papers show other links than annoyance and possibly sleep disturbance, and suggest that wind turbine visibility, negative attitudes, fear, or lobby groups cause adverse reporting, but provide no evidence dispelling impacts. Many argue acceptability of annoyance is a social measure set by government. Meantime, the Government of Canada "Wind Turbine Noise and Health Study" presented findings of an association between increasing noise from wind turbines and annoyance, but no found no evidence linking exposure to wind turbine noise to any of the self-reported illnesses, and no association between wind turbine noise and measures of stress, sleep quality, or significant changes in quality of life. Neglected in all of this, those who have been adversely impacted believe no one is listening. This paper takes up their position, to examine other acoustic factors to generate a hypothesis for cause of the human impact.

Contributed Paper

11:25

4aNS9. Annoyance of wind-turbine noise as a function of amplitude-modulation parameters. Christina Ioannidou, Sébastien Santurette, and Cheol-Ho Jeong (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, DTU Bygning 352, Kgs. Lyngby 2800, Denmark, ses@elektro.dtu.dk)

Amplitude modulation (AM) has been suggested as an important factor for the perceived annoyance of wind-turbine noise (WTN). Two AM types, typically referred to as "normal AM" and "other AM," depending on the AM extent and frequency region, have been proposed to characterize WTN AM. The extent to which AM depth, frequency, and type affect WTN annoyance is a matter of debate. In most subjective studies, the temporal variations of WTN AM have not been considered. Here, a sinusoidally modulated WTN model accounting for temporal AM variations was used to generate realistic artificial stimuli in which the AM depth, frequency, and type, while determined from real on-site recordings, could be varied systematically. Subjective listening tests with such stimuli showed that a reduction in AM depth, quantified by the modulation depth spectrum, led to a significant decrease in annoyance. When the spectrotemporal characteristics of the original far-field stimuli were included in the model and the temporal AM variations were taken into account by varying the modulation index over time, neither AM frequency nor AM type were found to significantly affect annoyance. These findings suggest that the effect of AM parameters on WTN annoyance may depend on the intermittent nature of WTN AM.

THURSDAY MORNING, 21 MAY 2015 — KINGS 1, 8:30 A.M. TO 12:05 P.M.

Session 4aPA

Physical Acoustics: Infrasound I

Roger M. Waxler, Chair
NCPA, University of Mississippi, 1 Coliseum Dr., University, MS 38677

Invited Papers

8:30

4aPA1. Incorporating atmospheric uncertainties into estimates of the detection capability of the IMS infrasound network. Alexis Le Pichon (CEA, DAM, DIF, Arpajon F-91297, France, alexis.le-pichon@cea.fr)

To monitor compliance with the Comprehensive Nuclear-Test-Ban Treaty (CTBT), a dedicated network is being deployed. Multi-year observations recorded by the International Monitoring System (IMS) infrasound network confirm that its detection capability is highly variable in space and time. Today, numerical modeling techniques provide a basis to better understand the role of different factors describing the source and the atmosphere that influence propagation predictions. Previous studies estimated the radiated source energy from remote observations using frequency dependent attenuation relation and state-of-the-art specifications of the stratospheric wind. In order to account for a realistic description of the dynamic structure of the atmosphere, model predictions are further enhanced by wind and temperature error distributions as measured in the framework of the ARISE project. In the context of the future verification of the CTBT, these predictions quantify uncertainties in the spatial and temporal variabilities of the IMS infrasound network performance in higher resolution, and will be helpful for the design and prioritizing maintenance of any arbitrary infrasound monitoring network.

8:50

4aPA2. Infrasound study of the atmospheric fine-scale wind velocity structure and its variability. Igor Chunchuzov, Sergey Kulichkov, Vitaly Perepelkin, Oleg Popov (Obukhov Inst. of Atmospheric Phys., 3 Pyzhevskii Per., Moscow 119017, Russian Federation, igor.chunchuzov@gmail.com), Jelle Assink (2CEA/DIF/DAM/DASE in Bruyères-le-Châtel, Arpajon, France), and Roger Waxler (3National Ctr. for Physical Acoust., University, MS)

The results of study of the fine-scale wind velocity structure in the upper stratosphere, mesosphere, and lower thermosphere by using recently developed method of infrasound probing of the atmosphere are presented. The method is based on the effect of infrasound scattering from highly anisotropic wind velocity and temperature nonhomogeneities in the middle and upper atmosphere. The vertical profiles of the wind velocity fluctuations in the upper atmosphere (up to a height of 140 km) are retrieved from the wave forms and travel times of the infrasound signals from volcanoes and surface explosions. The vertical wavenumber spectra of the retrieved wind velocity fluctuations are obtained for the upper stratosphere. Despite the difference in the location of explosive sources all the obtained spectra show the existence of high vertical wavenumber spectral tail with certain power law decay. The effect of a fine-scale wind velocity structure and its variability on the wave forms, coherence, and frequency spectra of the infrasound arrivals is studied. The possibility to use retrieved wind velocity structure in the upper stratosphere and lower thermosphere for improving an infrasound monitoring of infrasound sources in the atmosphere and parameterizing statistical characteristics of anisotropic turbulence (variances, spatial and temporal spectra, and coherence) is discussed.

9:10

4aPA3. Absorption of infrasound and acoustic-gravity waves in the atmosphere. Oleg A. Godin (Physical Sci. Div., CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., 325 Broadway, Mail Code R/PSD99, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

Long-range propagation of infrasound and acoustic-gravity wave fields in the middle and upper atmosphere are strongly affected by air viscosity and thermal conductivity. To characterize the wave dissipation, it is typical to consider idealized environments, which admit plane-wave solutions. This paper presents an asymptotic approach that relies instead on the assumption that spatial variations of environmental parameters are gradual. Unlike the traditional approach, the asymptotic theory allows one to derive AGW dispersion equations in a consistent manner for a wide range of scenarios and to describe the wave attenuation more realistically. Realistic assumptions about the atmosphere are found to lead to rather different predictions for absorption of atmospheric waves than the plane-wave solutions. Wind speed and wave frequency appear only through the intrinsic wave frequency in the dispersion equation, including the terms that describe the effects of viscosity and thermal conductivity. It is found that the Sutherland-Bass model of infrasound absorption should be modified to account for wind-induced absorption anisotropy, which arises from deviations of the intrinsic frequency from the wave frequency that is observed in a stationary reference frame. The anisotropy is expected to result in a significant decrease in the predicted attenuation of thermospheric returns.

9:30

4aPA4. Acoustic multipole source inversions of volcano infrasound. Keehoon Kim, David Fee (Geophysical Inst., Univ. of Alaska Fairbanks, 903 Koyukuk Dr. (Rm. 307A), Fairbanks, AK 99775, keehoon.kim@gmail.com), Jonathan M. Lees (Univ. of North Carolina Chapel Hill, Chapel Hill, Alaska), Akihiko Yokoo (Kyoto Univ., Kumamoto, Japan), and Mario C. Ruiz (Instituto Geofísico de la Escuela Politécnica Nacional, Quito, Ecuador)

Sources of volcano infrasound involve the atmospheric displacement associated with volcanic eruptions, where characteristic source dimensions are generally confined by the vent. Volcano infrasound sources are typically considered as a monopole which corresponds to the first-order term in the acoustic multipole expansion. However, when the wavelength becomes comparable to the size of the vent, the source may have further complexity which can be described only by higher-order terms, yet such complexity of volcano infrasound source has not been extensively explored. This is mainly due to (1) limited sampling of the acoustic wavefields due to poor network coverage and (2) complex sound propagation near the volcanic edifice which significantly distorts the multipole acoustic wavefields. In this study, we present a linearized waveform inversion technique incorporating numerical Green's functions. A full 3-D Finite-Difference Time-Domain (FDTD) method accelerated with GPU is used to compute accurate Green's functions taking into account volcano topography. The presented method is applied to infrasound data recorded at Sakurajima volcano (Japan) and Tungurahua volcano (Ecuador) and volcano infrasound sources associated with explosive eruptions are characterized in terms of a monopole and dipole. These methods could be applied to chemical explosions as well to determine source characteristics and complexity.

9:50

4aPA5. Spall effects on infrasound generation from explosions at the Nevada National Security Site. Kyle R. Jones (Sandia National Labs., 1515 Eubank Ave. SE, Albuquerque, NM 87123, krjones@sandia.gov), Arthur J. Rodgers (Lawrence Livermore National Lab., Livermore, CA), Rodney W. Whitaker (Los Alamos National Lab., Los Alamos, NM), Souheil M. Ezzedine, and Oleg Y. Vorobiev (Lawrence Livermore National Lab., Livermore, CA)

We apply two methods to evaluating the spall signature from underground chemical explosions such as those at the Source Physics Experiment (SPE) at the Nevada National Security Site (NNSS). The first approach uses the Rayleigh integral to compute overpressures for buried explosions from synthetic vertical acceleration data at surface ground zero. To obtain the acceleration data, we systematically vary parameters such as the spall duration, depth of burial and magnitude and observe the effect on the resulting acoustic waveform shape. The second method uses a hydrodynamic approach to more fully characterize the varied parameters to produce the acoustic waveforms. As the spall decreases we find that the acoustic waveform shape changes dramatically. This waveform signature may provide diagnostics on the explosive source and may be a useful metric for underground explosion monitoring. [This work was done under award number DE-AC52-06NA25946. Sandia National Laboratories is a multi-program laboratory managed and operated by Sandia Corporation, a wholly owned subsidiary of Lockheed Martin Corporation, for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-AC04-94AL85000.]

10:10

4aPA6. Infrasound from underwater sources. Läslo Evers (KNMI / TU Delft, PO Box 201, De Bilt 3730 AE, Netherlands, evers@knmi.nl), David Brown (CTBTO, Vienna, Austria), Kevin Heaney (OASIS, Lexington, MA), Jelle Assink (CEA, DAM, DIF, Arpajon, France), Pieter Smets (KNMI / TU Delft, De Bilt, Netherlands), and Mirjam Snellen (TU Delft, Delft, Netherlands)

Atmospheric low-frequency sound, i.e., infrasound, from underwater events has not been considered thus far, due to the high impedance contrast of the water-air interface making it almost fully reflective. Here, we report for the first time on atmospheric infrasound from a large underwater earthquake (Mw 8.1) near the Macquarie Ridge, which was recorded at 1325 km from the epicenter. Seismic waves coupled to hydroacoustic waves at the ocean floor, after which the energy entered the Sound Fixing and Ranging channel and was detected on a hydrophone array. The energy was diffracted by a seamount and an oceanic ridge, which acted as a secondary source, into the water column followed by coupling into the atmosphere. The latter results from evanescent wave coupling and the attendant anomalous transparency of the sea surface for very low frequency acoustic waves.

10:30–10:45 Break

10:45

4aPA7. Probabilistic infrasound propagation using ensemble based atmospheric perturbations. Pieter Smets and Läslo Evers (Seismology and Acoust., Royal Dutch Meteorological Inst., PO Box 201, De Bilt 3730 AE, Netherlands, smets@knmi.nl)

The state of the atmosphere is of utmost importance for infrasound propagation. In propagation modeling, the true state of the atmosphere is mainly represented by the analysis. The analysis is the best deterministic estimate of the atmosphere using a data assimilation system existing of a General Circulation Model (GCM). However, the analysis excludes error variances of both model and observations. In addition, the coarse resolution of GCM results in averaging of, e.g., clouds or gravity waves, over larger regions known as parameterization. Consequentially, arrivals due to fine-scale structure in wind and temperature can be missing. Therefore, infrasound propagation including the atmospheric best-estimate error variances based on an ensemble model is proposed. The ensemble system exists of model perturbations with an amplitude comparable to the analysis error estimates to obtain a probability density function. The best-estimate analysis error variances are described by a set of perturbations using the European Centre for Medium-range Weather Forecasts (ECMWF) Ensemble Data Assimilation (EDA) system. Probabilistic infrasound propagation is demonstrated by one year of mining activity, e.g., blasting, in Gällivare, northern Sweden, observed at infrasound array IS37 in Norway, part of the International Monitoring System (IMS) for verification of the Comprehensive Nuclear-Test-Ban Treaty (CTBT). Probabilistic infrasound propagation is compared with the standard deterministic result obtained using the analysis.

Contributed Papers

11:05

4aPA8. Infrasonic analysis of the October 28, 2014 Antares rocket failure at Wallops Island, Virginia, using video recordings as ground truth. Jay J. Pulli and Aaron Kofford (Raytheon BBN Technologies, 1300 North 17th St., Ste. 400, Arlington, VA 22209, jpulli@bbn.com)

We used close-in video recordings of the October 28, 2014 Antares rocket failure at Wallops Island, VA to establish an event timeline to aid in the analysis of infrasound recordings made at nearby stations of the IRIS Transportable Array. Our timeline is ignition at 22:22:38 UTC, liftoff at +4 s, bright plume and first explosion at an altitude of 300 m at +15 s, second large explosion as the rocket hits the ground at +25 s, followed by the excess fuel burn lasting some 400 s. Both explosions and the fuel burning events are seen in the infrasound data recorded at IRIS station S61A at a distance of 23 km, and the two explosions can be seen out to distances of 130 km. High resolution time frequency analyses of the infrasound signals at the distant stations show a dispersed signal from 0.5–8 Hz with a peak at 1.7 Hz and corresponding group velocity of 360 m/s. This dispersion curve corresponds to a low velocity duct at the surface with a thickness of approximately 1.2 km. The relatively fast group velocity can be attributed to the prevailing winds. Explosion yield estimates using the BOOM model indicate equivalent TNT yields of 20 and 200 tons for the two explosions. Colocated seismic and infrasound sensors at two stations allow us to estimate the acoustic-to-seismic spectral ratio at 1–10 μm/s/Pa. However, low coherence between the acoustic and seismic signals implies a non-linear transfer function at the sites.

11:20

4aPA9. Infrasound propagation and model reduction in randomly layered media. Christophe MILLET (CEA, DAM, DIF, Arpajon 91297, France, christophe.millet@cea.fr)

A consensus has emerged within the infrasound research community that gravity waves are filtered out in the available atmospheric models. Apart from occasionally strong lower-atmospheric effects, the major wave influences occur in the middle atmosphere, between 10 and 110 km altitudes. In the present approach, the unresolved gravity waves are represented as a random field that is superimposed on the average background state, and the wave equation is solved using a reduced-order model, starting from the classical normal mode technique. The reduced model is obtained by retaining a few propagating modes, with the aim of simplifying the acoustic model to the point that the predicted statistics of waveforms are correct, even though some small irrelevant details is lost. We focus on the asymptotic behavior of the transmitted waves in the weakly heterogeneous regime, for which the coupling between the wave and the medium is weak. The statistics of a transmitted broadband pulse are computed by decomposing the original pulse into a sum of modal pulses that can be described by a front pulse stabilization theory. Specifically, it is shown how reduced-order models can be used to explain some aspects of the variability of large infrasound datasets.

11:35

4aPA10. Sound induced plume instability. Konstantin A. Naugolnykh (Phys., Univ. of Colorado, 325 Broadway, Boulder, CO 80305, konstantin.naugolnykh@noaa.gov)

A sustained source of buoyancy creates a continuous rise of lighter fluid through the ambient denser fluid, with mixing occurring along the way. Such structure, called as a plume, is sensitive with respect to flow perturbations. In particular, the effect of sound can modulate the structure of plume as a result of sound-turbulent interaction. The acoustic wave can slightly change the structure of flow and then interact with in-phase spatially modulated turbulence. In application to plume this effect is considered in the present paper follow the approach of Chimonas, 1972 and Moiseev *et al*., 2000.

11:50

4aPA11. Hamiltonian ray tracing in an arbitrary curvilinear coordinate system—Amplitude estimation. Philip Blom and Stephen J. Arrowsmith (EES, Los Alamos National Lab., Los Alamos National Lab., PO Box 1663, Los Alamos, NM 87545, pblom@lanl.gov)

Acoustic ray tracing is known to be an efficient and robust numerical method to compute propagation paths of acoustic energy through the atmosphere and ocean. Over large propagation distances, a Cartesian formulation of ray tracing is known to be inadequate and a spherical or spheroidal formulation is required in order to obtain accurate propagation paths. An overview of the ray tracing equations in an arbitrary curvilinear coordinate system will be presented and compared with the known spherical coordinate formulation. Typically, coaxial ray paths are used to compute ray tube density and provide an estimation of the amplitude along the ray path. Here, the method of auxiliary parameters is used to identify the exact geometric spreading factor along individual ray paths. The implementation of the resulting ray tracing methods will be discussed including methods used to decrease computation time and planned additions to the propagation scheme.

THURSDAY MORNING, 21 MAY 2015 — KINGS 4, 8:00 A.M. TO 12:45 P.M.

Session 4aPP

Psychological and Physiological Acoustics and Speech Communication: Influence of Visual Cues on Auditory Perception

Jonas Braasch, Chair
School of Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

Chair's Introduction—8:00

Invited Papers

8:05

4aPP1. Understanding cross-modal interactions of spatial and temporal information from a cortical perspective. Barbara Shinn-Cunningham, Samantha Michalka, Abigail Noyce, and David Somers (Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu)

In the ventriloquism effect, perception of visual spatial information biases judgments of auditory events, yet there is little effect of auditory stimuli on perception of visual locations. In the flash-beep illusion, the number of auditory "beeps" biases judgments of the count of visual flashes, but visual flashes have little effect on perceived number of auditory events. These asymmetries suggest that vision is the "natural" sensory modality for coding spatial information, while audition is specialized for representing temporal information. Here, we review recent behavioral and neuroimaging evidence from our labs exploring the neural underpinnings of such perceptual asymmetries. Specifically, we find that there are distinct frontal cortical networks associated with visual information and auditory information. Yet these networks can be recruited by the other sensory modality, depending on task demands. For instance, when judging spatial aspects of auditory inputs, neural areas associated with visual processing are recruited; when judging temporal aspects of visual inputs, areas associated with auditory processing are activated. We also find another asymmetry: knowing *when* a spatial event is going to occur helps listeners judge location, but knowing *where* an event will occur does not help judgments about that event's timing. These kinds of studies help elucidate how temporal and spatial information is encoded in the brain, and the neural mechanisms by which visual-spatial and auditory-temporal information interact.

8:25

4aPP2. Localizing sound sources when the listener moves: Vision required. William Yost and Xuan Zhong (Speech and Hearing Sci., ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

In an article under review at JASA, we showed that when the listener rotates, determining the rotational direction of sound moving from loudspeaker to loudspeaker or if the sound is fixed at one loudspeaker location requires two forms of information: information provided the auditory spatial cues (e.g., interaural differences) AND information about the position of the head/body. Primary head/body position cues are provided by vision (and to some extent by the vestibular system and probably by other sensory and cognitive systems). In this paper, we address a very simple question which, as far as we know, has not been asked before: What is sound source localization accuracy when the listener moves? A sound source localization identification task was used to measure root-mean-square-errors (rms) in degrees in the full azimuth plane. Different stimuli were used and listeners were either rotated at a constant velocity or were stationary. They listened with their eyes open or closed. When listeners had no visual or others forms of information about head/body position (e.g., eyes closed) and, thus, all they had were the auditory spatial cues (e.g., interaural difference) sound source localization accuracy was very poor. [Research supported by an AFOSR grant.]

8:45

4aPP3. Visual capture of a stereo image. Yi Zhou and Christopher Montagne (Speech and Hearing Sci., Arizona State Univ., 975 S. Myrtle Ave., Coor 3470, Tempe, AZ 85287, yizhou@asu.edu)

This study investigated visual bias in localizing a "phantom" sound source generated by a stereo pair of speakers. The lateral position of the fused auditory image was controlled by varying the time delay or intensity ratio of two 15-ms noise bursts presented from two hidden loudspeakers positioned at +/− 45 degrees in azimuth. Visual stimuli were delivered from three LEDs positioned at −45, 0, and 45 degrees in azimuth. A multiple-alternative, forced choice task was implemented to measure listener's perceived sound localization with and without LED stimulation. The magnitude of visual capture (bias) was compared for the two stimulus manipulations—time delay and intensity ratio. Eleven normal hearing listeners participated in the task. Results showed stronger visual capture for time delay than intensity ratio manipulation. Binaural signal analyses were further conducted on ear-canal signals recorded from a KEMAR, showing that time delay manipulation introduced ambiguous binaural and spectral cues. The degree of visual bias was found to be correlated with the ambiguity of localization cues in the task.

9:05

4aPP4. Auditory/visual distance perception. Pavel Zahorik (Heuser Hearing Inst. and Div. of Communicative Disord., Dept. of Surgery, Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu)

A considerable body of research suggests that the perception of sound source distance exhibits systematic biases. The distances of far sources are progressively underestimated, but near sources are overestimated. Such biases are typically not observed in vision, where perceived distance is found to be highly accurate under natural viewing conditions in which a variety of visual distance cues are available. Relatively little is known about how distance information from both auditory and visual modalities is combined in the perception of distance, however. This is surprising, given that auditory/visual aspects of directional perception have been extensively studied, primarily in relation to the "ventriloquist effect." Here, two recent experiments on auditory/visual distance perception are summarized. The results from both suggest that not only is perceived distance less accurate in the auditory modality than in vision, but it is also considerably less precise. These results explain why visual information, when available, appears to dominate auditory information in the perception of distance. [Work supported by NEI.]

9:25

4aPP5. The interconnection of binaural and stereoscopic cues for distance perception. Kristen Murphy (Rensselaer Polytechnic Inst., 124 Fulton St. East, 2nd Fl., Grand Rapids, MI 49503, kmurphy@acousticsbydesign.com)

Encountering objects within our three-dimensional environment is a combination of sensory cues. While much research has been conducted on how the auditory and visual systems singularly locate objects in space, studies in cross-modal interaction of these two systems suggest the overall perception of a stimulus cannot fully be described by a single sensory system. The purpose of this study was to examine the relationship between visual and acoustic cues specifically in understanding depth perception. Previous studies of audio-visual interaction of spatial perception have accounted for many different auditory depth cues, but have not fully cross examined a number of visual depth cues. Participants were asked to determine the egocentric distance of a stimulus in a virtual three dimensional environment in terms of real-size quantitative scale. Stimuli were presented as either audio only, visual only, or audio and visual. The single-modal and multi-modal results were analyzed and compared against the null hypothesis that when presented simultaneously, the visual depth cues dominate the distance judgment and the acoustic depth cues have no bearing. In this study, the null hypothesis was not rejected, in accord with previous research investigating the proximity image effect.

9:45

4aPP6. Non-phonemic benefits of visual cues in the perception of consonants. Douglas Brungart, Lynn M. Bielski (Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.brungart@us.army.mil), and Eric R. Thompson (Air Force Res. Lab., Dayton, OH)

Visual cues from the talker's face improve speech perception because the talker adopts discrete facial configurations, known as visemes, corresponding to a limited number of possible phonemes in the auditory signal. Visual cues alone are insufficient for complete recovery of the speech signal, because individual visemes can occur with more than one phoneme. Visual speech cues may provide non-phonemic benefits in noise through segregation of target speech from background sounds. This experiment isolated phonemic and non-phonemic benefits of visual cues through identification of strings of consonants (i.e., aCHaBaGa) in an eight-alternative forced choice task. Listeners discriminated each consonant presented from a randomly selected foil in each position. In the heterovisemic case, the foil had a different viseme than the target. In the homovisemic case, each foil was from a list of consonants with the same viseme as the target, meaning the visual target provided no phonetic information about the speech. Results show benefit from visual speech cues in the homovisemic condition roughly half the size of that in the heterovisemic condition. This suggests a substantial portion of the visual benefit can be attributed to factors other than extraction of phonemic information from visemes in the visual stimulus.

10:05–10:15 Break

10:15

4aPP7. Individual differences in real-time processing of audiovisual speech by preschool children. Tina M. Grieco-Calub (The Roxelyn and Richard Pepper Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., FS, 2-231, Evanston, IL 60208, tinagc@northwestern.edu) and Janet Olson (School of Allied Health and Communicative Disord., Northern Illinois Univ., DeKalb, IL)

Visual speech cues maximize speech perception for adults in challenging listening environments: reliance on visual speech increases with greater auditory degradation. Young children, however, may be limited in audiovisual speech perception because of the cognitive costs associated with multimodal processing. In this study, real-time speech understanding was measured in preschool children ($N = 33$, 30–48 months of age) using eye-tracking methodology in the absence (quiet) or presence of two-talker babble and in the absence (auditory-alone, AO) or presence of audiovisual (AV) speech cues. On each trial, children were instructed by a female speaker to look at one of two objects projected onto a large screen. Speech processing was quantified by how quickly children fixated the target object (reaction time, RT) and overall accuracy of target-object fixation. Visual benefit was calculated as the difference in performance between the AO and AV conditions. Analyses revealed negative correlations between RT_{AO} and RT_{AV} in both the quiet and two-talker babble conditions: visual speech facilitated speech processing in children with slow RT_{AO}, but not in children with fast RT_{AO}. Results suggest that visual speech facilitates speech perception in preschool children, but is likely dependent on children's processing efficiency of the AO speech signal.

10:35

4aPP8. Auditory scene analysis with a multi-modal model. Thomas Walther (Institut für Kommunikationsakustik, Ruhr-Universität Bochum, Universitätsstraße 150, Bochum 44801, Germany, thomas.walther@rub.de)

By integrating state-of-the-art auditory scene-analysis methods with artificial cognitive feedback, the current project challenges human performance in quality of experience tasks and auditory scene analysis. Human listeners are regarded as multi-modal agents that develop their concept of the world by exploratory interaction. The prominent goal of the project is to develop an intelligent, computational model of active auditory perception and experience in a multi-modal context. The resulting system framework will form a structural link from binaural perception to judgment and action, realized by interleaved signal-driven (bottom-up) and hypothesis-driven (top-down) feedback processing within an innovative expert-system architecture. A conceptual overview of the project framework is presented, and insight is given into the current state of research, focusing on CASA-related search-&-rescue (S&R) scenarios. In these scenarios, an autonomous robot is endowed with auditory/visual feature-analysis facilities that provide it with bottom-up information of its environment. Top-down evaluation of the extracted features then results in cognitive feedback loops that allow the machine to adapt to complex S&R scenarios and perform significantly better than would be possible by only employing feed-forward control mechanisms. [Work performed in the context of the EU project Two!Ears.]

10:55

4aPP9. Considering, capturing, and representing the visual environment in noise perception studies. J. Parkman Carter (Architectural Acoust., Rensselaer Polytechnic Inst., 32204 Waters View Circle, Cohoes, NY 12047, cartej8@rpi.edu)

Despite studies which show the profound influence of visual environments on aural expectations and perceived noise annoyance, popular noise mapping techniques continue to reflect only one dimensional sound pressure level metrics. Contextual factors, which significantly influence noise perception, can be gleaned from the visual environment, but there are no established protocols for documenting and representing salient visual features. Techniques for capturing and representing the visual environment will be discussed, which can address the intrinsic differences between audio and photographic capture in terms of dynamic range and spatial resolution. Immersive audio and video projection techniques which are currently being developed at RPI's Collaborative Research Augmented Immersive Virtual Environment (CRAIVE) Lab will also be presented. These large scale multi-modal presentation strategies significantly improve efforts to conduct subject studies in environmental noise perception and assessment.

11:15

4aPP10. Visual influence on the subjective impressions of urban soundscapes. Tyler Adams (Architectural Acoust., Rensselaer Polytechnic Inst., 1402 N Mariposa Ave. #8, Los Angeles, CA 90027, echotyler@gmail.com)

Audio recording is a common method for evaluating subjective impressions of soundscapes, which is useful because the same information can be reproduced and presented to subjects in a controlled environment. This study was conducted to determine what impacts visual information might have in the subjective evaluations of urban soundscapes. Audio of urban environments was recorded using binaural and multi-channel methods; video was simultaneously recorded using a single channel digital camera. The soundscapes were reproduced for subjects in a laboratory environment with and without the accompanying videos. The data gathered from evaluations made in the laboratory were also compared with evaluations conducted *in situ* to determine the degree to which impressions might change in a controlled environment.

11:35

4aPP11. Defying the physical: Acoustic design as a medium for reimagining space and recontextualizing expression. Bobby E. Gibbs (Dept. of Commun. Sci. and Disord., Univ. of South Carolina, Columbia, SC 29208, artfull.mind@gmail.com) and Jonas Braasch (School of Architecture, Commun. Acoust. and Aural Architecture Lab, Rensselaer Polytechnic Inst., Troy, NY)

Just as the physical laws of light are both an inspiration and a point of departure for visual artists, the canvas of acoustic design is not aesthetically bound to real spaces. Using findings from venue visits, interviews and an interactive virtual auditory exploration, we will discuss the bricolage of physical space and cultural expression inherent in the dissemination of a unique musical genre. In particular, we will explore how experimental improvisers preserve and subvert spatial cues to create aural experiences that are at once intimate and illusive.

11:55

4aPP12. A bi-modal model to simulate auditory expectation for reverberation time and direct-to-reverberant energy from visual feedback. Jonas Braasch, M. T. Pastore, Nikhil Deshpande (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu), and Jens Blauert (Inst. of Commun. Acoust., Ruhr Univ. Bochum, Bochum, North Rhine-Westphalia, Germany)

A bi-modal model is presented that predicts the psychophysical results of Valente and Braasch [Acustica, 2008]. The model simulates expectation of room-acoustical qualities due to visual cues. The visual part of the model estimates angles of incidence and delays of the first two side reflections of a given frontal sound source. To this end, a stereoscopic image is used to determine azimuth angles and distances for the two frontal room corners. The distance estimates are derived by using the angular differences between the left- and right-eye images of each corner. The model then calculates the room volume by reconstructing a rectangular room from these data, assuming a range of possibilities for the missing room coordinates. In a next step, logarithmic fits of volume to expected reverberation time and of volume to direct-to-reverberant energy ratio predict the expected value ranges for these two parameters. Using a feedback structure, the visually-derived acoustic parameters become input to an auditory Precedence-Effect model, where they are used to pre-adjust inhibition parameters for two acoustic side reflections. These inhibition parameters are consequently refined in the course of the analysis of the incoming sound. [Work supported by NSF #1229391/#1320059 and ERC FP7-ICT-2013-C-#618075.]

Contributed Papers

12:15

4aPP13. Auditory illusions arising from lack of visual clues. Steven J. Waller (Rock Art Acoust., 5415 Lake Murray Blvd. #8, La Mesa, CA 91942, wallersj@yahoo.com)

Auditory illusions arising from lack of visual clues will be discussed and related to prehistoric cave art and Stonehenge. In the ancient past, when the wave characteristics of sound were not understood, virtual sound effects arising from complex sound wave interactions (echoes, reverberations, interference patterns, etc.) were misinterpreted as invisible beings (echo spirits, thunder gods, sound absorbing bodies, etc.) as described in ancient myths around the world. Scientifically conducted experiments involving blindfolded participants show how various ambiguous sounds can be interpreted in more than one way—like optical illusions. The blindfold in this case is a metaphor for ignorance of the wave nature of sound. It is proposed that ancient peoples were looking at the apparent source of sound and not seeing anything, and so concluded that the sounds emanated from beings that could not be seen. These experiments thus can help in understanding our ancestors' perceptions and reactions to sounds they considered mysterious and spooky. These discoveries are just a few examples of research findings that are springing from the new field of Archaeoacoustics. See https://sites.google.com/site/rockartacoustics/ for further examples.

12:30

4aPP14. Dynamic binaural sound source localization with ITD cues: Human listeners. Xuan Zhong, William Yost (Speech and Hearing Sci., Arizona State Univ., 975 S Myrtle Ave., Lattie F Coor Hall 2211, Tempe, AZ 85287, xuan.zhong@asu.edu), and Liang Sun (Dept. of Elec. and Comput. Eng., The Univ. of Texas at San Antonio, San Antonio, TX)

In real life, human listeners rarely experience cone of confusion errors in localization of sound sources due to the limitation of interaural time difference (ITD) as a spatial hearing cue. Previous work on robotics suggested that the confusion is disambiguated as successive observation of ITD is made over time, when self-motion of the microphone system was allowed. In this behavioral study, we investigated whether horizontal/vertical planes localization with time difference is possible when self-motion is allowed for human listeners. In particular, the case of a static sound source playing low frequency pure tone signal was studied. Human listeners were seated in a rotating chair in the middle of a loudspeaker array, and a low frequency audio signal was played at an elevation. The task was to identify the elevation spatial angle under conditions when vision was allowed and denied. The hypothesis was, with vision, a much more robust observation of self-motion is available, and the report of elevation would be more accurate. The results largely agreed with the hypothesis. A similar study for robotic hearing generated similar results.

Session 4aSA

Structural Acoustics and Vibration and Noise: Noise Identification and Control in the Mining Industry

Marcel Remillieux, Cochair
Los Alamos National Laboratory, Geophysics Group (EES-17), Mail Stop: D446, Los Alamos, NM 87545

Hugo E. Camargo, Cochair
Office of Mine Safety and Health Res., NIOSH, 626 Cochrans Mill Road, Building 155, Pittsburgh, PA 15236

Invited Papers

8:00

4aSA1. Noise identification, modeling, and control in mining industry. Debi Prasad Tripathy (Dept. of Mining Eng., National Inst. of Technol.,Rourkela, Odisha 769008, India, dptripathy@nitrkl.ac.in) and SANTOSH KUMAR NANDA (Ctr. for Res., Development and Consultancy (CRDC), Eastern Acad. of Sci. and Technol., Bhubaneswar, India)

Prolonged exposure of miners to the high levels of noise in opencast and underground mines can cause noise induced hearing loss and non-auditory health effects. To minimize noise risk, it is imperative to identify machinery noise and their impacts on miners at the work place and adopt cost effective and appropriate noise control measures at the source, path, and at the receiver. In this paper, authors have summarized the noise levels generated from different machineries used in opencast and underground mines and elaborated on frequency dependent noise prediction models, e.g., ISO 9613-2, ENM, CONCAWE, and non-frequency based noise prediction model VDI-2714 used in mining and allied industries. The authors illustrated the applications of innovative soft computing models, viz., Fuzzy Inference System [Mamdani and Takagi Sugeno Kang (T-S-K)], MLP (multi-layer perceptron, RBF (radial basis function) and adaptive network-based fuzzy inference systems (ANFIS) for predicting machinery noise in two opencast mines. The paper highlights the developments and research conducted on effective noise control measures being adopted for mining machineries and implemented in mines to minimize the noise menace so that noise levels generated in mines are within the prescribed noise standards and rules.

8:30

4aSA2. Comparison of noise reduction values for fit tests and work in coal mines. Brandon C. Takacs (Safety & Health Extension, West Virginia Univ., 3604 Collins Ferry Rd., Morgantown, WV 26506, brandon.takacs@mail.wvu.edu), Steven E. Guffey (Industrial Management Systems Eng., West Virginia Univ., Morgantown, WV), Mingyu Wu (Occupational Safety and Health Management, Grand Valley State Univ., Grand Rapids, MI), and Kevin Michael (Michael and Assoc., State College, PA)

At present exposure limits, one in four workers will develop a permanent hearing loss as a result of mining coal (Prince 1997). Mine Safety and Health Administration (MSHA) inspectors found in the period of 1986-1992 that approximately 25% of coal miners' daily noise doses exceeded MSHA's PEL. Virtually all mines have hearing conservation programs and virtually all miners are issued and told to wear either ear muffs or ear plugs. Nevertheless, miners still have a high rate of noise induced hearing loss (NIHL). An important question is to what degree the ineffectiveness of hearing conservation programs is due to failure of miners to wear muffs and plugs properly when they are needed and how much is due to inadequacies of hearing protectors. If the former is important, can technological innovations provide means to improve use of muffs and plugs. If the latter is important, can individual fit-testing improve noise reduction (NR) values achieved by miners. A related issue is whether fit-testing in an office environment adequately predicts NR values achieved during work if non-wearing times are excluded. To address those issues, WVU is conducting studies in a lab and in coal mines that primarily involve measuring sound levels in the ear (SPLear) concurrently with sound levels at the shoulder (SPLsh), allowing computation of NR values for protectors.

9:00

4aSA3. Calibration and assessment of noise monitoring instrumentation used by the Mine Safety and Health Administration. John P. Homer (Dept. of Labor, Mine Safety and Health Administration, 626 Cochrans Mill Rd., Pittsburgh, PA 15236, homer.john@dol.gov)

The use of noise dosimeters is the standard method for noise exposure monitoring and assessment in the United States mining industry. The Mine Safety and Health Administration (MSHA) calibrates and maintains a fleet of various noise dosimeters and acoustical calibrators for the purpose of conducting enforcement activities and providing technical support to mine operators. The dosimeter fleet is comprised of both corded- and badge-type instruments. MSHA studies have conclusively revealed evidence in support of both types as to performing acceptably both in practice and when tested against standardized criteria, ANSI S1.25-1991. MSHA operates an acoustical calibration laboratory which maintains National Institute of Standards and Technology (NIST) traceable calibration records for all noise dosimeters and calibrators used for enforcement and support functions. The laboratory is currently in the process of modernization. Replacement test systems will provide increased reliability, reduced turn-around time, and improved accuracy through the

implementation of modern computer-based components. Improvements provided by the laboratory modernization project will greatly improve quality assurance and provide the ability to pursue accredited compliance with ISO/IEC 17025:2005, through the American Association for Laboratory Accreditation (A2LA).

9:30

4aSA4. Vibration exposure characteristics and health risk prevention strategies associated with vibration exposure during mining applications. Tammy R. Eger (Human Kinetics, Laurentian Univ., 935 Ramsey Lake Rd., Sudbury, Ontario P3E 2C6, Canada, teger@laurentian.ca) and James P. Dickey (Kinesiology, Western Univ., London, Ontario, Canada)

Occupational exposure to vibration can lead to health problems and is typically classified as whole-body vibration (WBV), hand-arm vibration (HAV), or foot-transmitted vibration (FTV). Vibration exposure characteristics are typically measured and compared to standards (ISO 2631-1; ISO 5349-1; EU Directive 2002/44/EC) in an effort to determine the probability of adverse health effects including: low-back pain, spinal degeneration, and gastrointestinal tract problems linked with exposure to WBV; decreased grip strength, tingling/numbness in the fingers/hands, and blanching of the fingers associated with exposure to HAV; and numbness/tingling in the feet/toes, and cold induced blanching of the toes stemming from exposure to FTV. The presentation will summarize vibration exposure data collected by the research team over a 10-year period which suggests operators of load-haul-dump vehicles, haulage trucks, and dozers are exposed to WBV above recommended guidelines, while operators of jack-legs are exposed to HAV above guidelines. Our data also suggest miners exposed to FTV associated with drilling off of raise platforms and operating bolters and jumbo drills are at risk of developing a vibration-induced injury. The presentation will also review the effectiveness of control strategies evaluated by the research team including seating, isolation platforms, “anti-vibration” drills, cab interventions, road maintenance, and operating speeds.

10:00–10:15 Break

10:15

4aSA5. Measurement and analysis of noise and acoustic emission on a roof bolter for identification of joints and in rock. Jamal Rostami (Energy and Mineral Eng., The Penn State Univ., 126 Hosler Bldg., University Park, PA 16802, rostami@psu.edu), Soheil Bahrampour, Asok Ray (Pennsylvania State Univ., University Park, PA), and Craig Collins (R&D, JH Fletcher, Huntington, WV)

Interpretation of the data obtained from roof bolt drilling can offer a reliable source of information that can be used to characterize the ground for use in ground support design and evaluation. Drilling for installation of roof bolts is done by various drills and often it is a noisy process. The acoustic emission from the drill can be used as a source for identification of the rock formations and voids/joints in the rock mass. This paper offers a brief review of the ground support using roof bolts, followed by introduction and discussion of the roof characterization methods by instrumented roofbolters. In particular, installation of noise measuring sensors on the drill head to provide additional source of information will be discussed. The analysis and digital filtering of measured noises was used to identify the voids or joints in the full scale testing. A brief overview of the instrumentation and full scale testing, as well as review of the results of initial testing on an instrumented drills will be offered in the paper. The paper will introduce an algorithm for digital filtering of the acoustic emission to identify the discontinuities in the rock mass.

10:45

4aSA6. Field evaluations of a noise control for roof bolting machines. Amanda Azman (NIOSH, 626 Cochrans Mill Rd., Pittsburgh, PA 15226, aazman@cdc.gov)

Roof bolting machine (RBM) noise is a significant health hazard in underground coal mining because the sound levels at the operator’s station often exceed 100 dB(A). Drilling the hole prior to installing a roof bolt is the most significant source of RBM operator noise exposure. The dominant source of drilling noise is the drill steel. NIOSH has developed an enclosure to surround the drill steel as it is drills the hole to reduce the noise reaching the machine operator. The collapsible drill steel enclosure, or CDSE, is designed to collapse upon itself as the drill steel is further advanced into the mine roof so the drill steel can remain enclosed during the entire hole-drilling process. Original versions of this device effectively reduced noise exposure, but were deemed unacceptable by the mining community due to various usability and durability issues. A modified version has been developed and NIOSH laboratory tests revealed a 2–4 dB(A) noise reduction. This paper describes the in-mine evaluation of the redesigned CDSE. The field tests focused on the noise reduction as well as miner acceptance of the device. The results confirm improved usability as well as a 42% reduction in noise dose at the operator position.

11:15

4aSA7. Jackhammer chisel damper. Henry A. Scarton and Kyle R. Wilt (Mech., Aerospace, and Nuclear Eng., Rensselaer Polytechnic Inst., Rensselaer Polytechnic Inst., 110 Eight St., JEC 4008, Troy, NY 12180, scarton@rpi.edu)

This paper will present the performance of a rugged Coulomb Friction damper for a Jackhammer chisel. The predominant source of high intensity airborne noise from a pneumatically muffled jackhammer is the ringing resulting from impacting the undamped chisel, which radiates airborne sound from the impacts of the hammer exciting the transverse bending modes. A simulated steel chisel moil point was constructed with geometric properties similar to a jackhammer chisel and designed so as to survive the severe acceleration impacts from the reciprocating hammer. Anechoic tests of the chisel and it damped equivalent indicate that the sound pressure level for the undamped chisel due to longitudinal impact was 86.8 dB linear (re to 20 Pa) with the strongest ring tone at 1.6 kHz and harmonics; the sound pressure level for the damped chisel with identical axial impacts was reduced by 13 dB to 73.5 dB with severe reduction of chisel ring.

Session 4aSC

Speech Communication: Cross-language Speech Production and Perception (Poster Session)

Rachel M. Theodore, Chair
University of Connecticut, 850 Bolton Road, Unit #1085, Storrs, CT 06269

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

Contributed Papers

4aSC1. The role of articulatory cues in the establishment of perceptual categories. Emily Cibelli (Linguist, UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, ecibelli@berkeley.edu)

This study investigates the interaction of perceptual and articulatory information in the acquisition of non-native phoneme categories. There is evidence that perceptual learning can improve production of target sounds in a new language, but the reverse—improvement of perceptual discrimination from articulatory learning—is less well-studied (for one example, see Catford and Pisoni 1970). In this experiment, native English speakers learned Hindi coronal stop contrasts (dental vs. retroflex; 4-way VOT contrast) in a pre-test/training/post-test paradigm. Training involved production training, where subjects got explicit instruction about place of articulation and voicing of the target sounds, and practiced producing them. Comparison of pre- and post-test performance on an AX discrimination task indicated greater accuracy (β = 0.486, t = 2.763) and faster responses (β = −0.215, t = −3.396) for most target contrasts after training. (Those contrasts which did not improve tended to map onto English phonemic distinctions, and were well-discriminated at pre-test.) This result supports the hypothesis that articulatory information can contribute to the early development of novel perceptual categories under certain conditions. More generally, it suggests that information from one speech domain can be used to support representations in another speech domain.

4aSC2. Top-down linguistic categories dominate over bottom-up acoustics in lexical tone processing. Tian C. Zhao and Patricia K. Kuhl (Inst. for Learning & Brain Sci., Univ. of Washington, Portage Bay Bldg., Seattle, WA 98195, zhaotc@uw.edu)

Native tonal-language speakers exhibit reduced sensitivity to lexical tone differences within categories when compared to across categories (i.e., a top-down linguistic influence). However, increasing evidence suggests enhanced sensitivities to lexical tones among musically trained individuals (bottom-up influence), though previous studies examined non-tonal language speakers processing lexical tones. We investigated the relative contribution of top-down and bottom-up processing of lexical tones when both strategies are available. Seventeen native Mandarin speakers with extensive musical training completed a music test and a lexical tone discrimination task. The music test validated their enhanced pitch-processing abilities. The lexical tone discrimination task measured participants' lexical tone sensitivities along a continuum. Results were compared to existing data from Mandarin non-musicians and English speaking musicians (Zhao & Kuhl, 2012). Despite their enhanced bottom-up pitch processing, Mandarin musicians' performance exhibited patterns similar to Mandarin non-musicians, reflecting the dominant influence of linguistic categories ($p = 0.504$). Their sensitivities were also lower than English-speaking musicians ($p = 0.011$). However, further regression analysis suggests that bottom-up processing still plays a role in Mandarin musicians: their within-category tone discrimination is predicted by their musical pitch discrimination scores, similar to English musicians but opposite Mandarin non-musicians.

4aSC3. Mandarin Chinese vowel and tone identification in noise: Effects of native English experience. Mingshuang Li, Wenjing Wang, Sha Tao, Qi Dong (State Key Lab. of Cognit. Neurosci. and Learning, Beijing Normal Univ., Rm. 9417, Jingshi Hotel, No.19, Xinjiekouwai St., Haidian District, Beijing 100875, China, limingshuang@mail.bnu.edu.cn), Jingjing Guan, and Chang Liu (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX)

Several studies found differences in English vowel identification between Chinese-native listeners in the United States (CNU) and China (CNC) in multi-talkers babble (MTB) and temporally modulated (TM) noise, but not in quiet condition. Two possible explanations were proposed: first, CNU listeners used temporal modulation of noise more efficiently than CNC listeners; second, CNU listeners had less informational masking of MTB than their CNC peers. The current study aimed at exploring whether the difference in noise processing between CNU and CNC listeners was also presented for their native speech perception. Chinese vowel and tone identifications were measured for CNU and CNC in quiet, stationary and TM noise, babble-modulated noise and MTB. The identification scores of CNU listeners were significantly higher than CNC listeners in most noisy backgrounds, whereas both groups had the same performance in quiet. Moreover, compared with CNC listeners, CNU listeners gained greater masking releases from temporal modulation in noise at low SNRs, while the informational masking was comparable between the two groups. In conclusion, the 1–3 year residency in English-speaking country may improve Chinese-native listeners' capacity to use temporal cues in noise, but may not improve their ability against informational masking of English babble when listening to their native speech.

4aSC4. Cross-linguistic vowel variation in Saterland: Saterland Frisian, low German, and high German. Wilbert Heeringa, Heike Schoormann, and Jörg Peters (German Studies, Univ. of Oldenburg, P.O. Box 2503, Oldenburg 26111, Germany, wilbert.heeringa@uni-oldenburg.de)

This study investigates the vowel space of trilingual speakers of Saterland Frisian, Low German, and High German. The three vowel systems show differences in the number of distinct categories but share the majority of vowel qualities. To examine whether the size of the vowel space correlates positively with the number of vowel categories and whether distinctive vowels are positioned in the vowel space so as to increase perceptual contrast (Liljencrants and Lindblom, 1972, Lindblom, 1986) speakers were instructed to read vowels of all three languages in a /hVt/ frame. Measurements of mid-vowel F1 and F2 of monophthongs neither revealed a positive correlation between the size of the vowel space and the inventory size nor cross-linguistic differences of dispersion, except for a higher dispersion of High German vowels in the F2 dimension. Single vowel categories of Saterland Frisian and Low German were merged with respect to formant frequencies and duration. High German showed longer vowel duration and higher F2 of the front vowels than Saterland Frisian and Low German and a lower F1 than Low German. These results suggest that the trilingual speakers use the same phonetic categories for Saterland Frisian and Low German but not for High German.

4aSC5. Articulatory and acoustic correlates of English front vowel productions by native Japanese speakers. Sonya Mehta and William F. Katz (The Univ. of Texas at Dallas, 3531 Hilltop Ln., Plano, TX 75023, naya@utdallas.edu)

Speakers of languages having a relatively small vowel inventory (e.g., Japanese) may experience interference effects when attempting to produce novel vowels from a language with a larger vowel inventory, such as English. Studies of English vowels produced by native Japanese speakers have shown formant frequency and duration differences between native and accented productions. Less is known about the exact articulatory processes underlying second language vowel learning. The current study examines tongue position during native Japanese speakers' productions of American English vowels. Ten speakers produced the front vowels /i/, /ɪ/, /e/, /ɛ/, and /æ/ while seated in a 3D electromagnetic articulograph (EMA) system that tracked the position of the tongue (tongue tip, TT, tongue dorsum, TD, and tongue back, TB) and lower lip during vowel productions in the consonant environment /hVd/. Kinematic and acoustic measures were taken at the midpoint of each vowel steady-state portion. Differences in tongue position for tense-lax vowel pairs were determined by calculating the Euclidean distance between vowel centroids. Preliminary results suggest more spatial overlap in vowels produced by Japanese talkers than those produced by monolingual English speakers. The relationship between these articulatory distances and acoustic measures (formant frequency and durational changes) will be explored.

4aSC6. Mapping vowels from American English to Taiwanese Mandarin: A perceptual study. Yu-sheng Wang, Yueh-chin Chang, and Feng-fan Hsieh (Graduate Inst. of Linguist, National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu 30013, Taiwan, sam3300086@gmail.com)

It has been noted that vowel backness is largely preserved in English-to-Mandarin loanwords, but not vowel height. This asymmetry contradicts the native phonological patterns. The aim of this study is to re-examine this issue from an experimental perspective. Ten Taiwanese Mandarin (TM) speakers (aged 23–30) participated in this experiment. English Stimuli consisted in /CVmi/ sequences with C corresponding to one of the three consonants {b, d, g} and with V to one of the four vowels {ɛ, ɔ, æ, ɚ}. Mandarin stimuli also consisted in /CVmi/ sequences with C corresponding to one of the three consonants {p, t, k} and with V to one of the following vowels {i, a, u, ei, ie, au, ou, uo}. The participants were asked to rate similarity between American English and TM vowels (on scale 1–7). Our results partially support the generalizations in loanword adaptation, namely that TM speakers tend to map English [æ] onto Mandarin [ei], rather than [a], while [ɔ] is mapped to [ou/uo], [ɛ] to [ei], and [ɚ] to [ɤ]. We further measured the perceptual distance between English and Mandarin vowels by means of Euclidean distance. Given the assumption that smaller acoustic distance means greater perceptual similarity, it turned out that only this pair (English [ɚ] and Mandarin [ɤ]) can be explained away. It is thus concluded that loanword adaptation is not entirely based on raw acoustic signals. Phonological features and phonotactics also play a significant role.

4aSC7. On the fricative vowels in Suzhou Chinese. Fang Hu (Inst. of Linguist, Chinese Acad. of Social Sci., 5 Jian Guo Men Nei St., Beijing 100732, China, hufang@cass.org.cn) and Feng Ling (Shanghai Univ., Shanghai, China)

Apical vowels are widely distributed in Chinese dialects, whereas fricative vowels or strident vowels are less known. This paper is an acoustic and articulatory study of fricative vowels in the Suzhou dialect of Wu Chinese. The acoustic data from 20 speakers show that the fricative vowels have formant patterns between their plain and apical counterparts. Linguographic data from four speakers reveal that more laminal part of the tongue is involved in the production of the fricative vowels than their plain counterparts, which are basically anterodorsal. And the EMA study from three speakers confirms a comparatively advanced lingual configuration in the production of the fricative vowels. Although the production of fricative vowels is characterized by visible turbulent frication from the spectrograms, and a significantly lower harmonics-to-noise ratio, the results suggest that spectral characteristics of fricative vowels and apical vowels play a more important role in defining the vowel contrasts. That is, plain high vowels, fricative high vowels, and apical vowels distinguish in place of articulation, namely, being anterodorsal, laminal, and apical, respectively. And frication could be treated as a concomitant and redundant feature in the production of fricative or apical vowels.

4aSC8. Are opaque patterns what we think they are? The acoustics of the Bengali vowel chain shift. Traci C. Nagle and Kelly H. Berkson (Linguist, Indiana Univ., Memorial Hall 322, Bloomington, IN 47405, tcnagle@indiana.edu)

The chain-shift pattern of vowel height harmony in Bengali, in which low and mid vowels in monosyllabic verb stems alternate with mid and high vowels, respectively (æ~e, e~i, ɔ~o, o~u), is reported in the phonological literature and by Bengali linguists to be exceptionless. Experimental evidence, however, indicates that Bengali speakers extend this pattern to nonce verbs only about half the time when low vowels are involved. This raises questions about the productivity of the attested pattern, echoing the results of experimental tests of chain shifts in other languages (e.g., Polish, Taiwanese) in which speakers fail to extend opaque phonological patterns to nonce words. One question we may ask is whether the phenomenon described in the literature as an alternation truly neutralizes the vowel height contrasts in the Bengali language. This research presents instrumental acoustic analysis of Bengali vowels produced by native speakers in real and nonce verbs in order to address this question and gain new insight into both the quality of the vowels produced as a result of harmony in real words and how speakers apply the pattern with nonce verbs.

4aSC9. Temporal organization of frication in fricativized vowels. Matthew Faytak (Linguist, Univ. of California, Berkeley, 2632 San Pablo Ave., Apt. A, Berkeley, CA 94702, mf@berkeley.edu)

Fricative vowels are vocoids that exhibit a clear formant structure as well as fricative noise produced by a labiodental or (post)alveolar constriction (Connell 2007). *Apical vowels*, similar vocoids found in many Chinese languages, have known (post)alveolar constrictions, but it has been debated whether or not these actually generate substantial fricative noise (cf. Lee-Kim 2014). In order to better understand the relationship between these two segment types, acoustic data was obtained for the fricativized vowels in two Chinese languages (Standard Mandarin and Suzhou Wu) as well as Kom, a Grassfields Bantu language spoken in Cameroon. Analysis of fricativized vowel spectra and timecourses suggests two predominant types, one with steady-state fricative noise of a relatively low intensity, and a second that is more internally dynamic, exhibiting high-intensity frication toward the beginning of the vowel and lower-intensity frication towards the end. Standard Mandarin's apical vowels are shown to be broadly consistent with the second type, with less frication than typically described impressionistically. However, Suzhou Wu and Kom speakers vary between the two types. This suggests that the language-specific phonetic implementation of this fricative noise does not neatly align with any commonly used descriptive terms.

4aSC10. Phonation type contrasts and vowel quality in Marathi. Kelly Berkson (Dept. of Linguistics, Indiana Univ., 1021 E. Third St., Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu) and Stephen Politzer-Ahles (New York Univ., Abu Dhabi, Abu Dhabi, United Arab Emirates)

Phonation-type contrasts in consonants and vowels are typically associated with a number of acoustic differences. Phonemic breathy phonation, for instance, is fairly consistently associated with lower Cepstral Peak Prominence values than modal sounds, and with higher values for spectral measures such as H1-H2 and H1-A3. The effect of breathy phonation on vowel quality, however, is less consistent: lower first formant (F1) values have been found for breathy vowels in some languages, while others do not show consistent F1 differences based on phonation type. Furthermore, at present little work has investigated the effect of breathy voiced consonants on formant values in subsequent vowels. The current study presents data from Marathi, an Indic language with phonemically breathy-voiced obstruents and sonorants. Ten native speakers (five males and five females) produced real words that included the vowels [a] and [e] after both modal and breathy consonants. The effect of consonant phonation type on F1 and F2

values in subsequent vowels is reported in order to augment our understanding of the relationship between voice quality differences and vowel quality.

4aSC11. Persian speakers of English: Acoustics of vowel epenthesis. Christina C. Akbari (Commun. Disord., Arkansas State Univ., PO Box 910, State University, AR 72467, cakbari@astate.edu), Katsura Aoyama (Commun. Disord., Univ. of North Texas, Denton, TX), and James Dembowski (Speech, Lang., and Hearing Sci., Texas Tech Health Sci., Lubbock, TX)

This study results from the need to develop further understanding into the process of vowel epenthesis which is often observed with second language learners. The purpose of this study was to examine the epenthetic vowels produced by Persian speakers of English to determine the acoustical characteristics and to ascertain if these vowels were acoustically different or similar to the main vowels. Past research indicated that at times the vowels could be copies of the main vowels but at other times they were different. However, no acoustical data were provided in regards to the acoustical characteristics of the epenthetic vowels (durations, F1, and F2 frequencies). Twenty Persian speakers of English took part in the study. All participants arrived in the United States after the age of 22. All of the participants demonstrated limited English proficiency on a standardized measure. The results indicated that the epenthetic vowels produced were qualitatively different from the main vowels in terms of duration, F1, and F2 frequencies.

4aSC12. Vowel context effects on the spectral dynamics of English and Japanese sibilant fricatives. Patrick F. Reidy and Mary E. Beckman (Dept. of Linguist, The Ohio State Univ., 24A Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, reidy.16@osu.edu)

Previous analyses of vowel context effects on the sibilant fricatives of English (/s, ʃ/) and Japanese (/s, ɕ/) have focused on spectral properties computed from a limited number of time points, e.g., frication midpoint or vowel onset. However, the spectra of sibilants vary temporally; thus, it is worth considering how their spectral dynamics vary across vocalic contexts. Vowel context effects were investigated with respect to the trajectories of three psychoacoustic measures computed across the timecourse of native, adult speakers' productions of word-initial, pre-vocalic sibilants. Psychoacoustic spectra were computed from 10 ms windows, spaced evenly across the frication, by passing each window through a bank of gammatone filters, which modeled the auditory system's differential frequency selectivity. From these psychoacoustic spectra, the peak frequency, excitation-drop (difference between maximum high-band and minimum low-band excitation), and the half-power bandwidth of the peak were computed. In Japanese, the peak frequency and the excitation-drop trajectories showed effects of vowel height—the trajectories for high (vs. mid and low) vowel contexts diverging from 50–75% of the fricative duration onward. In English, the excitation-drop trajectories showed similar effects of vowel height; however, the trajectories for high vowels diverged later than in Japanese. Peak bandwidth exhibited context effects only for Japanese /ɕ/, where it was lower in back vowel contexts across the first 75% of the fricative duration.

4aSC13. Phonetic context effects in second-language phonetic category discrimination. Dave C. Ogden (Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan Ave., Ann Arbor, MI 48109, ogdend@umich.edu)

In an exemplar-based model, effects of linguistic experience in L2 acquisition are explained as the conspiracy of experienced tokens in reshaping phonetic category representations (Goldinger, 1998; Johnson, 1997). Such a model predicts that acquisition of a new category is facilitated in phonetic contexts, which are highly dissimilar to L1 experience. This prediction was tested for English-speaking learners' acquisition of the prevoicing-short lag bilabial stop contrast in French. In a phoneme-monitoring task, French learners heard French CV words and responded whenever they heard "the sound p." Word-initially, [p] exemplifies the category /b/ in English but /p/ in French. French vowels included English-like [i u e o] and front rounded non-English [y ø]. If learners' stop category representations retain experienced stops in context, stop tokens preceding [y ø] fall only in the French pre-voicing [b] to short-lag [p] VOT range, while stops preceding [i u e o] include tokens of both languages' categories ([b], [p], and [p^h]). This should lead to faster identification of unaspirated [p] as an instance of /p/ before French-only [y ø] than before shared vowels. Identification of [p] as /p/ was significantly faster before French-only vowels; representations of phonetic segments appear to include information about phonetic context.

4aSC14. Cross-linguistic spectral and voicing properties of [v]. Christina Bjorndahl (Linguist, Cornell Univ., 203 Morrill Hall, 159 Central Ave., Ithaca, NY 14853-4701, cjm295@cornell.edu)

This study reports on the spectral characteristics and voicing properties of [v] in English, Greek, Russian, and Serbian, specifically testing whether inventory structure and the phonological status of [v] correlate with phonetic differences. Phonologically, [v] patterns with obstruents in Greek and English, with sonorants in Serbian and with both obstruents and sonorants in Russian. The present study expands upon previous work in two ways. First, it compares the results to English, which differs from the other three languages in having [w] against which [v] contrasts. Second, it tests the hypothesis that spectral differences correlate with the degree of devoicing. Our results show that [v] is fully and consistently voiced throughout its duration in Greek, Russian, and Serbian, regardless of environment, but exhibits significant devoicing in all environments in English, and moreover varies greatly across speakers. These results show first that the correlation between the spectral properties and phonological status of [v] in Greek, Russian, and Serbian does not arise due to differences in the degree of devoicing of [v]. Second, they suggest that when phonological status is the same, as in Greek and English, other factors such as inventory structure may interact with the phonetic realization of a segment.

4aSC15. The production of Korean coronal obstruents by inexperienced English-speaking learners of Korea. Hanyong Park (Linguist, Univ. of Wisconsin-Milwaukee, Curtin Hall 523, P.O. Box 413, 3243 N. Downer Ave., Milwaukee, WI 53211, park27@uwm.edu)

English-speaking learners of Korean often experience difficulty in learning Korean laryngeal categories (i.e., fortis, lenis, and aspirated). We examine whether the degree of difficulty in producing these laryngeal categories generalizes across the whole coronal obstruent system (i.e., stops and fricatives) and two different prosodic locations (i.e., CV and VCV). In our experiment, English-speaking college students taking first semester Korean course read a list of Korean frame sentences with target stimuli. The target stimuli consist of three coronal stops /t t' t^h/ and two coronal fricatives /s s'/ combined with the vowel /a/ in CV and VCV. To assess production accuracy, native speakers of Korean are asked to identify the consonants from the learners' productions. Results indicate that lenis categories are the most difficult across stops and fricatives in the two prosodic locations, suggesting that the learners develop production skills for laryngeal contrast applicable across the entire obstruent system. However, the degree of difficulty patterns are not similar in CV and VCV: the learners were more accurate producing /t^h/ in VCV than in CV. Thus, these findings suggest that the learners develop the production skills separately for different prosodic locations.

4aSC16. Articulatory targets for coronals in Taiwan Mandarin: A study of EMA, palatography, and linguagraphy. Wei-rong Chen and Yuen-chin Chang (National Tsing Hua Univ., 2F-5, No.62, Ln. 408, Zhong-Hua Rd., Zhubei City, Hsinchu County 302, Taiwan, waitlong75@gmail.com)

Coronal consonants hold a special status for their crowded space for articulatory contrasts. Mandarin and its variants are known to have rich inventories of coronal consonants, where a three-way coronal place contrast is usually maintained. Nonetheless, previous studies have reported a (partial) neutralization of coronal places of articulation in Taiwan Mandarin, while other acoustic studies contend that the three-way contrast remains. The aim of this study is to evaluate the purported coronal neutralization from an articulatory perspective. Five native speakers of Taiwan Mandarin participated in a set of electromagnetic articulographic (EMA), palatographic and linguagraphic experiments in order to investigate the articulatory targets of coronal consonants in Taiwan Mandarin. Our articulatory results are in general consistent with the acoustic results in previous studies, in that the three-way coronal place contrast does exist in Taiwan Mandarin, and the so-called "retroflexes" in Taiwan Mandarin were produced with

constrictions being at "alveolar" zone, rather than "postalveolar" zone as in Beijing Mandarin.

4aSC17. Articulatory similarity in rhotic sounds: A cross-linguistic comparison. Suzanne Boyce, Sarah M. Hamilton, Ahmed Rivera Campos, and Varsha Nair (Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., 345b, Cincinnati, OH 45267, boycese@ucmail.uc.edu)

The question of what phonetic quality defines rhotics as a natural class has been debated. Clinical reports indicate that in many languages, rhotics are developmentally late to emerge and subject to errors that are resistant to remediation, suggesting that rhotics may be distinguished by "complexity in articulation." In American English, the complexity of articulation may derive from its doubly articulated nature; simultaneous palatal and pharyngeal tongue constrictions are consistent features across variants. For other languages with a rhotic liquid, however, it is unclear if this pharyngeal constriction gesture is also an articulatory feature. Using ultrasound, this study compares rhotics from different language families to describe the presence or absence of pharyngeal constriction. Results indicate that pharyngeal constriction is observed across languages in rhotic allophones that have been described as difficult for children to acquire.

4aSC18. A kinematic analysis of Japanese /r/: Effects of syllable position and vowel context. William F. Katz (Commun. Sci. and Disord., Callier Ctr. for Commun. Disord., Univ. of Texas at Dallas,1966 Inwood Rd., TX 75235, wkatz@utdallas.edu), Sonya Mehta (Commun. Sci. and Disord., Univ. of Texas at Dallas, Plano, TX), and Amy Berglund (Commun. Sci. and Disord., Univ. of Texas at Dallas, Dallas, TX)

Japanese /r/ is claimed to vary substantially as a function of context. Greater stricture is thought to occur after pauses and at initial syllable position when compared with intervocalic position. The environment of certain vowels is thought to result in changes to the degree of lateralization of /r/. However, few studies have investigated the articulatory processes involved in realizing the varieties of Japanese /r/ production. To address this issue, we obtained speech recordings for ten native talkers of standard Japanese using an Electromagnetic Articulography (EMA) system. Each talker produced 12 repetitions of /ra/, /ri/, /ru/, /re/, /ro/ in a carrier phrase designed to contrast syllable boundaries: /sore wa VCV VCV desu # CV/ (e.g., "This is *ara ara* [pause] *ra*"). Kinematic data were recorded with sensors placed on the tongue tip (TT), tongue body (TB), tongue left lateral (TL), tongue right lateral (TR), and lower lip (LL) positions. Results to date suggest syllable-initial (post-pausal) /r/ has the greatest stricture, followed by the first VCV production. Productions also showed vowel-dependent differences, with /a/ and /o/ showing noticeable lateral movement patterns. The kinematic data will be compared with perceptual ratings of /r/ quality by native English speakers.

4aSC19. Acoustic and articulatory characteristics of "weak" and "strong" initial sibilants in Chamalal (Andic). Sven Grawunder (Dept. of Linguist, Max Planck Inst. for Evolutionary Anthropology, Deutscher Platz 6, Leipzig 04103, Germany, grawunder@eva.mpg.de) and Zaynab Alieva (Dagestanian State Pedagogical Univ., Makhachkala, Russian Federation)

In the phonological descriptions of a number of Caucasian languages appears a "weak" vs. "strong" contrast for otherwise voiceless obstruents, which was previously described as a lenis-fortis contrast, but recently attributed as geminate. Hence, we investigate field and lab data from the Gigatli dialect of Chamalal (CJI), a language belonging to the Andic branch of Avar-Andic-Tsezic group in the Nakh-Dagestanian family, spoken in Dagestan (Russian Federation). The weak-strong contrast in Chamalal involves glottalic and pulmonic fricatives at the same place of articulation. Concretely the series of central alveolar sibilant fricatives (/s/, /sː/, and /sː'/) is focused, of which all can occur in initial position. Sagittal ultrasound-articulatography of one female speaker demonstrates wide consistency in place of articulation of the tongue tip, although differences in pre-dorsal curvatures between /sː'/ and /s, sː/ are observed. The acoustic data are based on word list elicitations of six speakers (two males/four females): the ejectives show pronounced central envelope peaks and highest intensity slopes of the following vowel, but take a middle position in duration. Two distant peaks are frequently observed in the longer envelopes of "strong" /sː/. /s/ and /sː'/ differ strongest for (>2kHz) weighted COG, whereas /sː/ ranges widely in between.

4aSC20. Dental, or retroflex, that is the question: A study of Mina stop consonants. Alexandra Abell and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, alabell@indiana.edu)

Mina (also commonly known as Gen) is a language spoken in Togo and the Mono department of Benin. Mina remains little studied in comparison to the related language Ewe. The present study attempts to shed light on Mina phonetics. In particular, the question whether Mina voiced coronal stops form a single category or two categories distinguished by retroflexion is investigated through acoustic, articulatory, and perceptual analyses.

4aSC21. Acoustic evidence for incomplete neutralization of coda obstruents in Korean. Kyounghee Lee, Ann Bradlow, and Matt Goldrick (Linguist Dept., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, kyoungheelee2013@u.northwestern.edu)

The phonetic manifestation of phonological contrast neutralization is of long-standing interest for our understanding of the phonetic realization of phonological structure. Korean phonology is particularly interesting in this regard with three dimensions of coda contrast neutralization: laryngeal, manner, and palatal. Extending previous work (Kim & Jongman, 1996) that documented complete neutralization of manner contrasts, the present study investigated all of the three types of coda neutralization in both spontaneous (orthography absent) and read speech (orthography present). The results revealed no acoustic differences between most pairs of coda obstruents subject to manner or palatal neutralization. However, laryngeal neutralization was phonetically incomplete with lenis codas (/p, t, k/) differing from their non-lenis counterparts (/p^h, t^h, k'/). In spontaneous speech, this was observed in preceding vowel duration, voicing into closure, and closure duration; in read speech, voicing into closure, closure duration, and release burst duration of a following consonant differed. Our new findings demonstrate that laryngeal coda neutralization in Korean is phonetically incomplete regardless of the presence or absence of orthography. As a case of neutralization beyond final devoicing, this study extends the empirical basis of linguistic theory, discriminating between the predictions of traditional theories versus more recent ones such as exemplar-based theory.

4aSC22. An acoustic study of pulmonic and ejective stops in the final coda position in Yucatec. Masaki Noguchi (Linguist, Univ. of Br. Columbia, 4449 West 14th Ave., Vancouver, British Columbia V6R2Y2, Canada, mskngch@alumni.ubc.ca)

In this study, I present an acoustic analysis of contrast between pulmonic and ejective stops in the phrase final coda position in Yucatec, a Mayan language. Ejective consonants are found in about 16% of the world's language, and there are a relatively large number of acoustic studies about these sounds. However, a significant empirical shortcoming of the previous studies is their exclusive focus on the acoustic realization of ejective consonants in the syllable onset position. This is partly due to a cross-linguistic tendency that ejective consonants are prohibited to occur in the syllable coda position. Therefore, data from Yucatec, which allows ejective consents to occur in the syllable coda position, fill the empirical gap. This study compared pulmonic stop /k/ and ejective stop /k'/ in the phrase final coda position taking a number of acoustic measurements. The results showed (1) stop release is shorter for the ejective, (2) stop closure duration is shorter for the ejective, (3) stop release noise intensity is higher for the ejective, and (4) the contrast between pulmonic and ejective stops affects the voice quality of the preceding short vowel (H1-H2 is lower for the short vowel preceding the ejective).

4aSC23. Uptalk in Southern Californian English and Mexican Spanish. Karla Y. Barranco Márquez (Linguist, UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0108, kbarranc@ucsd.edu)

"Uptalk" is the tendency for declarative sentences to end with rising intonation instead of falling intonation. In Southern Californian English, uptalk is realized with a shallow pitch rise [L-H% in Tones and Break Indices (ToBI) labeling]. In contrast to this are polar questions, which are typically realized with steep rises (H-H%) [Ritchart and Arvaniti 2014]. Uptalk has also been impressionistically found in the variety of Spanish spoken in Mexico City. Mexican Spanish has three forms of phrase-final rises: low-to-high rises (LH%), low-to-mid rises (LM%), and high-to-high rises (HH%) [de-la-Mota *et al.* 2010], whereas English only has low and high rises [Beckman & Ayers Elam 1997]. Given the larger inventory of possible phrase-final melodies in Spanish vs. English, uptalk may be realized differently in the former than the latter. To determine whether uptalk in Spanish differs from uptalk in English, we analyzed the intonation from one representative female speaker of each language. Fifty phrases from each language were extracted from recordings of two television shows, *Laguna Beach* (for English) and *Rebelde* (for Spanish). Two coders then labeled each phrase using ToBI conventions for English and Spanish. The phonological and phonetic differences between English vs. Spanish uptalk will be discussed.

4aSC24. Sentence recognition in temporal modulated noise for native and non-native listeners: Effect of language experience. Jingjing Guan, Chang Liu (Univ. of Texas at Austin, 2504A Whitis Ave., Austin, TX 78712, jane.guan@utexas.edu), Sha Tao, Mingshuang Li, Wenjing Wang, and Qi Dong (State Key Lab. of Cognit. Neurosci. and Learning, Beijing Normal Univ., Beijing, China)

Speech recognition in adverse listening conditions is more difficult for non-native listeners than native listeners. Previous work in our laboratories found that Chinese-native listeners with native English exposure may improve the use of temporal cues of noise for English vowel identification. The purpose of this study was to investigate whether such benefits of using temporal modulation in noise were also presented in sentence recognition. IEEE sentence recognition in quiet, stationary, and temporally modulated noise were measured for American English native (EN) listeners, Chinese-native listeners in the United States (CNU), and Chinese-native listeners in China (CNC). Results showed that in general, EN listeners outperformed the two groups of CN listeners in quiet and noise, while CNU listeners had better scores of sentence recognition than CNC listeners. Moreover, at low SNRs, the masking release on sentence recognition from the temporal modulation in noise was greatest for EN listeners, and the smallest for CNC listeners with the CNU listeners in between, while at middle and high SNRs, there was no significant group effect on the masking release, consistent with the findings of English vowel perception. The group difference in using temporal modulation of noise may be associated with acoustic differences between Chinese and English speech.

4aSC25. Listener sensitivity to English- and Spanish-specific sounds in a language categorization task. Cynthia P. Blanco (Linguist, Univ. of Texas at Austin, 305 E. 23rd St., B5100, Austin, TX 78712, cindyblanco@utexas.edu), Colin Bannard (Psych., Univ. of Liverpool, Liverpool, United Kingdom), and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX)

Listeners are able to distinguish native from non-native speech with as little as 30 ms of input (Flege, 1984; Kondaurova & Francis, 2008), suggesting that listeners' mental representations include fine-grained acoustic detail about typical and atypical pronunciations in their native language. The current study considers the relative contribution of different language-specific sounds to listeners' language categorization. English monolinguals and early and late Spanish-English bilinguals categorized nonce words containing a language-specific target segment as belonging to English or Spanish. Target segments included the English phonemes /θ, ɹ/, the Spanish phoneme /r/, and the Spanish and English versions of /l, u/, which exist in both languages but vary in their phonetic implementation. All listeners accurately categorized stimuli with /ɹ, r/ and the Spanish pronunciations of /l, u/. English /θ/ and /l, u/ variants received more mixed responses. For these more difficult sounds, late bilinguals were more accurate than the other groups. Early bilinguals responded faster than the other groups to stimuli with language-specific phonemes, and both bilingual groups responded more quickly than monolinguals to /l, u/. The results reveal that the bilinguals' experience with both languages enhances their sensitivity to language-specific cues and may lead to more detailed sound representations for their languages.

4aSC26. Word learning by naïve non-native speakers may not be predictable by cognitive test scores. Irina A. Shport (English, Louisiana State Univ., 260-G Allen Hall, Baton Rouge, LA 70803, ishport@lsu.edu)

The role of cognitive abilities in learning of words with two types of novel sounds—lexical tones and coronal consonants—was compared. Individual differences were expected to play a larger role in implicit learning of tones than consonants. Twenty-two English speakers took tests assessing working memory, attention, executive function, and processing speed (NIH & Northwestern University, 2006–2012, *NIH Toolbox: Cognition*). Then, they learned twelve Vietnamese words varying in their initial consonants [m, t, ʈ, z̨] and tones (high level, falling, and rising). After training with feedback, the learning was assessed in a 12-word alternative identification test ($N = 792$). The range of accuracy scores was 11.1–80.6%, with the mean accuracy of 43.6% ($SD = 18.5$). The proportion of tonal errors was significantly larger than the proportion of consonantal errors [$\chi(1) = 174.68$, $p < 0.001$]. Mixed-effect regression analyses showed, however, that cognitive test scores did not explain the variance in the word identification accuracy. These results suggests that although lexical tones are more challenging than consonants for speakers of non-tone languages, adult beginning learners may be able to learn words with different types of novel sounds equally efficiently, regardless of individual differences in cognitive abilities.

4aSC27. The comparison of auditory capacity between bilinguals and monolinguals by consonant–vowel dichotic test. Lina Motlagh Zadeh (Commun. Sci. and Disord., Univ. of Cincinnati, 3239 Bishop St., Apt #4, Cincinnati, OH 45220, motlagla@mail.uc.edu), Leila Jalilvand Karimi (Audiol., Shahid Beheshti Medical Sci. Univ., Tehran, Iran), and Noah Silbert (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

The purpose of the present study is investigation of differences in auditory capacity between bilinguals and monolinguals using the consonant–vowel (CV) dichotic perception test. Bilingualism is one well-studied in psychology and linguistics, with a number of studies revealing that exposure to two languages can lead to the changes in the central system. Auditory capacity refers to the ability to relay information in sound patterns to higher brain centers, with observed measures of auditory capacity reflecting the maximum amount of information that can be processed by the auditory system. The present study probed auditory capacity in 80 normal individuals, 40 bilinguals and 40 monolinguals. Members of the bilingual groups spoke either Turkish or Kurdish from birth and began learning Persian as a second language at or before 6 years of age. Listeners identified distinct consonants presented to each ear in a CV dichotic perception test. Consonant identification accuracy served as a measure of auditory capacity of individual listeners. Results indicated that auditory capacity was greater in the bilingual group. In general, higher scores were gained by bilinguals relative to monolinguals. Due to the large number of bilinguals, knowledge of the similarities and differences between bilinguals and monolinguals and investigation of the effects of second language on central auditory processing are important.

4aSC28. Accounting for multicompetence and restructuring in the study of speech. Charles B. Chang (Dept. of Romance Studies, Boston Univ., 621 Commonwealth Ave., Boston, MA 02215, cbchang@post.harvard.edu)

Phonetic studies meant to generalize to monolingual speakers of a target language have often examined individuals with considerable experience using another language, such as the immigrant native speaker. This paper presents, first, results from a meta-analysis of the literature, suggesting that conflation of ostensibly bilingual ("multicompetent") individuals with monolinguals remains common practice and, second, longitudinal data on speech production that demonstrate why this practice is problematic. Adult native English speakers recently arrived in Korea showed significant changes in acoustic properties of their English production during their first weeks of learning Korean ("phonetic drift") and, furthermore, continued to show altered English production a year later, months after their last Korean class and without extensive use of Korean in daily life. These patterns suggest that the linguistic experience associated with residence in a foreign language environment tends to induce and then prolong phonetic drift of the native language, making the multicompetent native speaker living in a foreign language environment unrepresentative of a monolingual in the native language environment. The speed and persistence of these effects highlight the need for language researchers to be explicit about the population under study and to accordingly control (and describe) language background in a study sample.

4aSC29. Audience design in non-native speech. Jenna T. Conklin, Ashley Kentner, Wai Ling Law, Mengxi Lin, Yuanyuan Wang, and Olga Dmitrieva (Linguist Program, Purdue Univ., 100 North University St., West Lafayette, IN, wlaw@purdue.edu)

Speech has been shown to accommodate the communicative needs of listeners, for example, for increased intelligibility compared to normal speech. Previous research shows that native speakers adapt their speech in the presence of noise (Garnier *et al*., 2006), and when addressing children (Biersack *et al*., 2005) or foreigners (Scarborough *et al*., 2007). However, our knowledge of how *non-native* speakers modify their speech depending on the interlocutor is limited. The goal of this study is to identify acoustic features of non-native speech, which may be affected by a change in listener characteristics, particularly, in terms of language background. Native speakers of Mandarin from the same dialectal area gave directions in a map task to three confederate participants: another native speaker of Mandarin (a non-native speaker from the same L1 background), a native speaker of English (a native speaker of participants' L2), and a native speaker of Russian (a non-native speaker from a different L1 background). Acoustic variables associated with audience design in speech, including measures of speech rate, pitch, vowel duration, vowel quality, and vowel space were examined and compared to results of a detailed survey of participants' language experience and attitudes toward their first and second languages.

4aSC30. Phonetic drift in a first language dominant environment. Wendy Herd, Robin Walden, Whitney Knight, and Savana Alexander (MS State Univ., 2313 Lee Hall, PO Box E, MS State, MS 39762, wherd@english.msstate.edu)

Phonetic drift, changes in the first language (L1) sound system as a result of acquiring a second language (L2), has been documented in learners immersed in L2-dominant environments. Less attention has been given to phonetic drift in speakers learning an L2 in L1-dominant environments. In this study, we conducted a cross-sectional analysis of English learners of Spanish in the United States at beginning ($N = 12$), intermediate ($N = 12$), advanced ($N = 9$), and near native ($N = 6$) proficiency levels. Participants were recorded reading a pseudo-randomized list of words including *heed*, *hayed*, *who'd*, *hoed* to measure drift in vowels and *poll*, *bowl*, *toll*, *dole*, *coal*, *goal* to measure drift in oral stops. Significant differences in vowel quality and in the VOT of voiced stops were found. Intermediate, advanced, and near native learners of Spanish produced vowels in more peripheral positions of the vowel space than beginning learners of Spanish. Similarly, intermediate, advanced, and near native learners produced voiced stops with more negative VOTs than beginning learners. All of these effects were strongest in near native learners. These results suggest that phonetic drift occurs not only when learners are immersed in L2-dominant environments but also as a result of language instruction in L1-dominant environments.

4aSC31. Adaptation to accent affects categorization, but not basic perceptual representation. Matt Lehet and Lori L. Holt (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Baker Hall, Pittsburgh, PA 15213, mil@andrew.cmu.edu)

Listeners rapidly reweight the mapping of acoustic cues to speech categories in response to local changes in short-term input [Idemaru & Holt, 2011]. For instance, when encountering an accent that reverses the typical correlation of acoustic cues to speech category membership. Here, we examined the level at which this rapid learning occurs. Listeners incidentally encountered an artificial accent in Blocks 1 and 3 of a vowel categorization task. In these blocks, the typical relationship between spectral quality and duration was reversed for /ɛ/ and /æ/ (*setch, satch*) such that /ɛ/ had longer durations than /æ/. Participants down-weighted their reliance on the duration of spectrally ambiguous vowels after brief exposure to the accent. They rapidly returned to using duration when the English correlation was restored (Block 2). Additionally, we tested whether vowel duration exerts a durational contrast effect on adjacent consonants and, if so, whether this effect was modulated by perceptual down-weighting of duration in Blocks 1 and 3. We observed consistent durational contrast effects across blocks. Even when participants down-weighted vowel duration for categorization, vowel duration exerted a durational contrast effect on the following consonant. Learning leaves the basic perceptual representation intact. We describe a model of the results.

4aSC32. Contributions of practice with feedback and testing without feedback to learning of a non-native phonetic contrast. Beverly A. Wright, Emma K. LeBlanc, Jessica S. Conderman, and Courtney S. Coburn (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60202, b-wright@northwestern.edu)

Listeners learn non-native phonetic contrasts given practice and feedback, but what is the dose-response curve for that training? Does testing without feedback affect that response? We trained native English speakers on a non-native phonetic contrast between pre-voicing and voicing on two consecutive days, varying the number of daily training trials across groups. Practice with feedback for 240, 120, or 60 trials/day all led to a similar steepening of the category-boundary slope, but practice with feedback for 30 trials/day yielded no learning. To determine whether the daily pre- and post-training tests without feedback (120 trials/test) affected this outcome, we trained another group 60 trials/day with feedback, but removed all but the final post-test. This group did not learn, indicating the tests without feedback facilitated improvement. Finally, to establish whether the benefit from these tests came from task performance, we trained another group 60 trials/day with feedback, but replaced the original tests with stimulus exposure alone until the final post-test. This group's ending performance resembled that obtained with task performance during the tests without feedback. Thus, trials with feedback can function as an all-or-none trigger for recruiting the contribution of trials without feedback, or mere stimulus exposures, to speech learning. [NIH supported.]

4aSC33. Neurophysiological correlates of perceptual learning of Mandarin Chinese lexical tone categories: An event-related potential study. Guannan Shen and Karen Froud (Teachers College, Columbia Univ., 509 West 122nd St., Apt. 18, New York, NY 10027, mandy.g.shen@gmail.com)

Recent studies have shown that certain brain event-related potentials (ERPs) are sensitive to auditory perceptual categorical boundaries. This study investigates brain responses to lexical tone categorization for three groups of adult listeners: (1) native English speakers who had no exposure to Mandarin before age 17, but took advanced Mandarin courses as adults; (2) naïve English speakers; and (3) native Mandarin speakers. Two tonal continua were derived from natural speech through interpolation within two tonal contrasts (Tone 1/Tone 4; Tone 2/Tone 3). First, category boundaries were examined through classic identification and discrimination tasks. Secondly, high-density electroencephalography (EEG) was used to record brain responses while participants listened to tones in two oddball paradigms: across-category and within-category. If perception of lexical tones is categorical, cross-category deviants are expected to elicit larger ERP responses (specifically, mismatch negativity (MMN) and P300) than within-category deviants. Both behavioral and ERP results indicate that lexical tones are perceived categorically by native Chinese speakers but not by inexperienced English speakers. Although English learners of Chinese demonstrated categorical perception in behavioral tasks, their ERP response amplitudes were attenuated, and did not differ between within- and across-category conditions. Acoustic cues and characteristics of L2 phonological learning in adulthood are discussed.

4aSC34. Asymmetries in the perception of Mandarin tones: Evidence from mismatch negativity. Stephen Politzer-Ahles and Kevin Schluter (NYUAD Inst., New York Univ. Abu Dhabi, PO Box 903, New York, NY 10276-0903, spa268@nyu.edu)

The mismatch negativity (MMN) component of the electroencephalogram reflects phonological asymmetries: specifically, greater MMN is elicited by infrequent (deviant) tokens when the features of the frequent (standard) token are phonologically underspecified than when those features are phonologically fully specified. Such asymmetries have not been studied in non-Indo-European languages or in suprasegmental contrasts, which might not be represented in the same way as segments. Therefore, the present study investigated the neural representation of Mandarin contour tones. Mandarin third tone (T3) is realized as second tone (T2) in certain contexts, whereas it has no alternation relationship with fourth tone (T4). The present study examined MMNs elicited by T3~T2, T3~T4, and T2~T4 contrasts (in both directions). Asymmetrical MMN effects were elicited in both T3~T2 and T3~T4 contrasts: MMN was smallest when the standard was T3. The results suggest that T3 has a less specified neural representation than both T2, which it alternates with, and T4, which it does not. This may be due to the featural representation of T3 (in standard Mandarin it is a Low tone, which is typologically less marked) or due to its behavior (i.e., it may be the case that features which alternate always trigger reduced MMN effects).

4aSC35. Asymmetries in vowel perception: Effects of formant convergence and category "goodness". Matthew Masapollo, Linda Polka (McGill Univ., 2001 McGill College, 8th Fl., Montreal, Quebec H3A 1G1, Canada, matthew.masapollo@mail.mcgill.ca), and Lucie Ménard (Univ. of PQ at Montreal, Montreal, Quebec, Canada)

The mechanisms underlying directional asymmetries in vowel perception have been the subject of considerable debate. One account—the Natural Referent Vowel (NRV) framework—suggests that asymmetries reflect a *language-universal* perceptual bias, such that listeners are predisposed to attend to vowels with greater formant convergence (Polka & Bohn, 2011). A second (but not mutually exclusive) account—the Native Language Magnet (NLM) theory—suggests that asymmetries reflect an experience-dependent *language-specific* bias favoring "good" exemplars of native language vowel categories (Kuhl, 1993). We tested the above hypotheses by investigating whether listeners, from different language backgrounds, display asymmetries influenced by formant proximity and/or language experience. Specifically, we examined monolingual English and French listeners' performance in a within-category AX vowel discrimination task, using variants of /u/ that systematically differed in *both* their degree of formant proximity (between F1 and F2) and category "goodness" judgments. Results revealed asymmetries that pattern as predicted by NRV when pairs of /u/ tokens exhibited a relatively larger difference in their F1-F2 convergence patterns, and as predicted by NLM when the pairs of /u/ tokens exhibited a relatively smaller difference in their F1-F2 convergence patterns. These findings suggest that language-universal perceptual biases and specific language experience interact to shape vowel perception.

4aSC36. Nonnative phonetic category training in varying acoustic environments. Eleni L. Vlahou, Aaron Seitz (Psych., Univ. of California, Riverside, 900 University Ave., Riverside, CA 92521, evlahou@gmail.com), and Norbert Kopčo (Inst. of Comput. Sci., P. J. Šafárik Univ., Košice, Slovakia)

Past research has shown that reverberation has a pronounced detrimental effect on speech intelligibility, but no previous studies investigated how it affects the acquisition of novel phonetic categories. Here, we present a follow-up to a previous study that examined adult nonnative phonetic learning using speech stimuli presented in reverberant environments [Vlahou *et al.* (2014), ARO Abstract #806]. Listeners were trained to discriminate a novel dental-retroflex contrast in Hindi. Using virtual acoustics, stimuli were presented in anechoic space, in a single room, or in multiple rooms. For some subjects, training was supervised, consisting of a 2AFC categorization task with trial-by-trial feedback. For other subjects, training was unsupervised, in the form of a videogame which promoted stimulus-reward contingencies. Performance was evaluated on trained and untrained sounds presented in familiar and unfamiliar rooms. Overall, trained listeners outperformed untrained listeners. Supervised training induced more robust learning of the trained material and generalization of learning to untrained voice. Introducing multiple reverberant environments was beneficial for unsupervised, but not for supervised, training. These results suggest that phonetic adaptation to reverberation is robust and that experiencing novel phonemes in various acoustic environments can enhance unsupervised phonetic category learning. [Work supported by APVV-0452-12, TECHNICOM ITMS: 26220220182, and NSF-BCS-1057625.]

4aSC37. English listeners use suprasegmental lexical stress online during spoken word recognition. Alexandra Jesse (Psychol. and Brain Sci., Univ. of Massachusetts, Amherst, MA), Katja Poellmann, and Ying-Yee Kong (Commun. Sci. & Disord., Northeastern Univ., 360 Huntington Ave., 226 FR, Boston, MA 02115, yykong@neu.edu)

Lexical stress facilitates spoken word recognition. In English, listeners primarily rely on vowel reduction in unstressed syllables as a cue to lexical stress, but are nevertheless sensitive to suprasegmental cues (Cooper *et al.*, 2002 *Lang. Speech*). In the present study, we tested with a visual-world eye-tracking paradigm whether English listeners utilize suprasegmental stress cues online during word recognition. On each trial, listeners heard one of four displayed printed words. Two of the words (critical words) were segmentally identical in their first two syllables but differed in the suprasegmental realization of stress in their first syllable. The first syllable had either primary (e.g., admiral) or secondary (admiration) lexical stress. The other two words were phonologically and semantically unrelated to the critical items. In the critical word trials, listeners looked more frequently at the targets with initial primary stress than at the competitors with secondary stress during the presentation of the first two syllables. No difference was found when the targets had secondary stress. The degree to which competitors were fixated was not modulated by stress. This suggests that English listeners use the presence, but not the absence, of suprasegmental cues to primary lexical stress during word recognition.

4aSC38. Rule-based and information-integration categorization during an incidental learning task. Casey L. Roark and Lori L. Holt (Psych. Dept., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, croark@andrew.cmu.edu)

Research in visual category learning supports the existence of dual category-learning systems with distinct neurobiological substrates. Rule-based category learning is engaged by category input distributions that vary orthogonally in perceptual space whereas information integration learning is engaged when categories are defined across multiple input dimensions. Investigators have recently begun to explore these dual systems in audition, including phonetic category learning. However, investigations of dual category learning systems have tended to use category training tasks in which participants actively search for category-relevant dimensions, make overt category labeling decisions, and receive explicit feedback about the correctness of their responses. Recent neuroimaging research highlights the fact that overt category training tasks like this engage learning systems that differ from those engaged by more incidental category learning. Since most natural category learning occurs under more incidental conditions, it is important to understand how training tasks interact with category learning systems. The aim of the present research is to investigate the interaction of input distribution and task. We examine nonspeech auditory category learning across the same perceptual input space for rule-based and information-integration categories in incidental and overt training tasks to characterize how training task interacts with input category distributions.

4aSC39. Listeners' sensitivity to nasal coarticulation and its interactions with lexical neighborhood density. Kuniko Nielsen (Linguist, Oakland Univ., 320 O'Dowd Hall, Rochester, MI 48309-4401, nielsen@oakland.edu) and Rebecca Scarborough (Linguist, Univ. of Colorado Boulder, Boulder, CO)

Previous research has shown that words from dense phonological neighborhoods, thus subject to greater lexical competition, are hyperarticulated (Wright, 2004; Munson and Solomon, 2004) and produced with a greater degree of coarticulation (Scarborough, 2013). The current study investigates

listeners' sensitivity to lexically conditioned degree of nasal coarticulation. If the basis of attested Neighborhood Density (ND) effects is listener-oriented (as suggested by Wright), listeners' sensitivity to different degrees of coarticulation may be influenced by ND patterns as well. Forty-three native speakers of American English participated in a forced-choice discrimination task. The stimuli consisted of 32 monosyllabic words with nasal codas, and the degree of vowel nasality was natural or artificially increased or decreased. A Generalized Linear Mixed Effects regression was performed on Correct Response with Direction of Manipulation and ND as fixed factors. The model revealed a significant interaction of Direction and ND ($p < 0.001$), where increased nasality was discriminated more correctly among high ND words while decreased nasality was discriminated more correctly among low ND words. This result shows that listeners are indeed sensitive to differences in degree of nasal coarticulation, and in ways that reflect neighborhood conditioned patterns.

4aSC40. Perceptual scaffolding of non-native speech categories through videogame-based training. Ran Liu and Lori L. Holt (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, ranliu@cmu.edu)

Adults learn many new tasks with ease, but acquiring the sounds of a new language is notoriously difficult. Decades of attempts to develop effective training regimens have focused primarily on highly *explicit* approaches to training adults to categorize non-native speech sounds. Participants are aware of the phonetic distinctions they are learning, focus their attention on the contrasts or categories of interest, and are provided with some form of overt trial-by-trial feedback. A growing body of perceptual learning research suggests that directing attention away from the task of learning sounds by embedding training in tasks with overt goals that are unrelated to sound training can lead to efficient learning gains. We leverage this idea to incidentally train native English-speaking adults on non-linguistic auditory categories that are highly relevant to Mandarin Chinese lexical tone perception. We observed significant incidental learning of the nonspeech auditory categories as a result of five days of incidental videogame training and highly significant generalization to natural spoken productions of Mandarin tones. We also observed preliminary evidence that nonspeech category learning transferred to boost Mandarin vocabulary learning. We discuss these results in the context of incidental category learning and perceptual scaffolding of non-native speech categories.

4aSC41. Incidental auditory category learning. Lori L. Holt, Yafit Gabay (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15232, loriholt@cmu.edu), Frederic Dick (Birkbeck, Univ. of London, London, United Kingdom), and Jason Zevin (Univ. of Southern California, Los Angeles, CA)

Little is known about how auditory categories are learned incidentally—that is, in the absence of overt category decisions or experimenter-provided feedback. This is an important gap, as learning in the natural environment does not arise from explicit feedback and there is evidence that the learning systems engaged by traditional tasks are distinct from those recruited by incidental category learning. We examined incidental auditory category learning with a novel paradigm, the Systematic Multimodal Associations Reaction Time (SMART) task, where participants rapidly detect and report the appearance of a visual target in one of four possible screen locations. Although the overt task is rapid visual detection, a brief sequence of sounds precedes each visual target. These sounds are drawn from one of four distinct sound categories that predict the location of the upcoming visual target. These many-to-one auditory-to-visuomotor correspondences support incidental auditory category learning. Participants incidentally learn categories of complex acoustic exemplars and generalize this learning to novel exemplars and tasks. Further, learning is facilitated when category exemplar variability is more tightly coupled to the visuomotor associations than when the same stimulus variability is experienced across trials. We relate these findings to theories of incidental phonetic category learning.

Session 4aSP

Signal Processing in Acoustics: Beamforming, Optimization, Source Localization, and Separation

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Contributed Papers

9:00

4aSP1. Information-theoretic optimization of multiple-sensor positioning for passive narrowband acoustic source localization. Thomas J. Hayward and Mitchell A. Potter (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Fundamental information-theoretic quantities, including conditional entropy of source location given received complex spectral values and per-iteration information gain (relative entropy) are applied as performance metrics to the optimization and real-time adaptation of receiver array spatial configurations for iterative (sequential) Bayesian localization of narrowband acoustic sources. Computational examples illustrate the application of these performance metrics to the adaptation of mobile-sensor spatial positioning and to the optimization of element positions for fixed vertical arrays in a shallow-water waveguide. Evolutionary search algorithms [Bäck and Schwefel, Evolutionary Computation **1**(1), 1–23 (1993)] are investigated as a unified computational approach to both optimization problems. The optimized array spatial configurations are compared with configurations optimized with respect to traditional (energy-based) performance metrics, and the differences are interpreted. [Work supported by ONR.]

9:15

4aSP2. On the estimation of bearing using a single hydrophone. Edmund Sullivan (Res., prometheus, 46 lawton brook Ln., Portsmouth, RI 02871, bewegungslos@fastmail.fm)

It is well known that a single moving hydrophone can estimate the bearing of a pure tone source if the source frequency is known *a priori*. It is also well known that with two or more hydrophones, a moving array can provide an improved estimate of bearing by jointly estimation the source frequency and bearing with a Kalman filter. This is due to the introduction of the phase information contained in the Doppler. This leads to the question of whether by jointly estimating the bearing and source frequency with a Kalman filter, a single hydrophone will suffice. By using a form of the observability matrix adapted to the nonlinear case, it is shown that in general, it is not possible. The reason is shown to be related to the fact that the bearing information in the Doppler is not sufficient and the introduction of a spatial phase, by introducing at least one more hydrophone is necessary.

9:30

4aSP3. The effect of ground sensor and UAV flight path geometry on the tomographic reconstruction of a weakly sheared daytime convective planetary boundary layer. Anthony Finn and Kevin Rogers (Defence & Systems Inst., Univ. of South Australia, Bldg. W, Mawson Lakes, SA 5095, Australia, anthony.finn@unisa.edu.au)

A signal analysis of the sound of a propeller-driven unmanned aerial vehicle (UAV) shows that its acoustic signature comprises a set of strong narrowband tones superimposed onto a broadband random component. If such a UAV overflies an array of microphones, the projected and observed Doppler shifts in frequency of the narrowband tones may be compared and converted into effective sound speed values: 2- and 3D spatially varying atmospheric temperature and wind velocity fields may then be estimated using tomography. The technique has practical application in a number of research fields. In this paper, we examine the influence of UAV flight path and ground sensor geometry on the feasibility and usefulness of UAV-based atmospheric tomography. Realistic conditions for a weakly sheared daytime convective atmospheric boundary layer are synthesized through use of massively parallel large eddy simulation code that utilizes pseudo-spectral differencing in horizontal planes and solves an elliptic pressure equation. Particular attention is paid to the accuracy with which the surface layer (lowest 50 m of atmosphere) may be reconstructed using UAV-based acoustic tomography as this region typically experiences the greatest spatio-temporal variation in temperature and wind speed; and arrangements of UAV flight path and sensor geometry do not permit ray paths to intersect without the UAV flying very low and disturbing the atmosphere. The influence of meteorological observations obtained onboard the UAV and by ground sensors is also examined.

9:45

4aSP4. Azimuth-elevation direction finding using one four-component acoustic vector-sensor spread spatially along a straight line. Yang SONG (Dept. of Elec. Eng. and Information Technol., Universität 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)

A considerable literature has developed to use the four-component acoustic vector-sensor to estimate the azimuth-elevation direction-of-arrival of incident acoustic sources. A four-component acoustic vector sensor (a.k.a. a "vector hydrophone" in underwater applications) consists of a pressure sensor, plus three identical but perpendicular univariate velocity-sensors, which are often idealized as collocated. One shortcoming of the above acoustic vector sensor is the point-size of its array aperture, due to the spatial collocation of all its four component-sensors. If these four component-sensors are placed at different spatial locations, there would arise spatial phase factors among these component-sensors' data, and the array-manifold of (1) would be invalid, thus the incident wavefield's propagation direction is no longer so straight-forwardly obtainable. However, Song & Wong (JASA **4**, 2013) shows that how the four component-sensors may be allowed to spread out arbitrarily in three-dimensional space (thus sampling the incident wavefield at different locations), while not just achieving direction finding, but to do so with increased accuracy. This paper will focus on one particular array grid to spread out the four component-sensors—on a straight line in any permutation.

10:00–10:15 Break

10:15

4aSP5. Role of adaptive beamforming with random and irregular arrays. Paul Hursky (HLS Res. Inc., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

When we design an array to accommodate as many as 4–5 octaves of bandwidth, the design must provide a variety of apertures and element spacings. Further, the concept of operations may be such that no structure is available for maintaining a regular array structure (even if we actually wanted one). One example of this is an array of freely drifting sonobuoys. Clearly, with a limited number of elements (far fewer than would be needed to provide half wavelength spacing for the highest frequency over the aperture needed at the lowest frequency), compromises must be made. Solutions include combinations of nested apertures, geometrically increasing spacings, and random spacings to varying degrees—in short irregular and random arrays. Such designs have very poor conventional beam response patterns, with very high sidelobes, compared to the fully populated arrays spaced at half wavelength. However, these objections are a red herring. We will show that adaptive beamforming produces beam responses with very low sidelobes. We will also show that irregular and random arrays indeed outperform regular arrays (of equivalent cost), if a wide bandwidth and range of interference directions are considered. We will also briefly review requirements for array calibration for adaptive beamforming.

10:30

4aSP6. Compressive beamforming with LASSO. Peter Gerstoft (SIO Marine Phys. Lab. MC0238, Univ. of California San Diego, 9500 Gillman Dr., La Jolla, CA 92093-0238, gerstoft@ucsd.edu) and Angelliki Xenaki (Appl. Mathematics and Comput. Sci., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

For a given sound field observed on an array, compressive beamforming reconstructs the direction-of-arrival (DOA) map using a sparsity constraint on the source distribution. The problem is posed as an underdetermined problem expressing the pressure at each receiver as a superposition of plane waves associated with each DOA, a phase delayed sum of source amplitudes. The L1 sparsity constraint makes the problem solvable with convex optimization and the sparsity constraint gives improved resolution. We here derive the sparse source distribution using maximum a posteriori (MAP) estimates for both single and multiple snapshots. Compressive beamforming does not rely on any matrix inversion and thus works well even for single snapshots where it gives higher resolution than conventional beamforming. For multiple snapshots in a free-space environment, compressive beamforming has a performance similar to MVDR. In a multi-path environment MVDR fails, but compressive beamforming works well. The superior resolution of compressive beamforming is demonstrated with vertical array data from event S5 in the SWellEx96 experiment in a multi-path shallow water environment.

10:45

4aSP7. On beyond hidden source estimates—Improving the output of blind source separation algorithms. Richard Goldhor, Joel MacAuslan, and Keith Gilbert (Speech Technol. & Appl. Res., 54 Middlesex Turnpike, Entrance D, Bedford, MA 01730, rgoldhor@sprynet.com)

In the presence of multiple simultaneously active acoustic sources, microphone response signals are typically additive mixtures of "acoustic images"—that is, filtered and delayed versions of hypothetical yet intangible "source signals." Under certain conditions, Blind Source Separation (BSS) algorithms can "demix" sets of such microphone responses into output signals often called "Hidden Source Estimates" and characterized as estimates of those inaccessible source signals. However, despite the name of these algorithms and the characterization of their outputs, BSS algorithms generally do not actually reconstruct the original source signals. Indeed, it is not even clear that the notion of "source signal" is always well-defined. We argue that a valid BSS output can best be understood as an estimate of some unknown member of an entire equivalence class of related signals, and that certain members of these classes are much more useful and ontologically well-grounded than others. Notably, acoustic images are particularly interesting members of these equivalence classes. We present a method for converting arbitrary hidden source estimates into acoustic images, and explore the utility of enhancing BSS algorithms so that they output specific acoustic images rather than generic hidden source estimates.

11:00

4aSP8. Blind source localization in a room based on wavefield separation. Thibault Nowakowski, Julien De Rosny, Laurent Daudet (Institut Langevin - UMR7587 - CNRS, 1 rue Jussieu, Paris 75005, France, thibault.nowakowski@espci.fr), and François Ollivier (UPMC - Institut Jean le Rond D'Alembert, St-Cyr l'Ecole, France)

Narrowband localization of sources in a room is a challenging problem because of the multiple reflections off the walls. Recently, we have been developing methods to localize monochromatic sources within an array of a few tens of microphones, without any knowledge on the physical properties of the room. To that end, we use a wavefield dereverberation approach in which the diffuse part is canceled, thanks to a projection operator built from a plane wave basis. However, this basis requires that no heterogeneities are present in the space of interest between the microphones. To overcome this limitation, in the case of an heterogeneous space, we show that a new projection operator can be experimentally built from a set of measurements of the responses between the microphones and some sources. This projection operator removes the reverberation, and can be used as preprocessing to locate the sources despite the heterogeneities, for instance by using classical beamforming processing. The method is first validated with numerical simulations. Then, experiments are performed in a large room, with an array composed of 100 microphones. A source, emitting at 500 Hz, can be located close to a strong reflector with an accuracy of about 10 cm.

11:15

4aSP9. Blind separation of heart sounds. Lingguang Chen, Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, fj6467@wayne.edu), Yong Xu (Dept. of Electric and Comput. Sci. Eng., Wayne State Univ., Detroit, MI), William D. Lyman, and Gaurav Kapur (Dept. of Pediatrics, Wayne State Univ., Detroit, MI)

This paper presents blind separation of heart sounds by de-convoluting the directly measured mixed signals using the temporal Green's function or the point source separation (PPS) method (Wu and Zhu, JASA, 2012). The objective of this study is to examine the feasibility of using a practical and cost-effective method to separate the aortic, pulmonic, mitral and tricuspid sounds that are involved in the first and second heart sounds, respectively. It is emphasized that because many parameters such as the locations from which the heart sounds are originated, as well as the speed at which heart sounds travel inside a human body are unknown a priori, it is unrealistic to expect a perfect separation of the heart sounds. To begin with, we conduct a numerical simulation test that uses an iteration scheme to locate sources and determine the speed of sound. This is done by applying PPS algorithm repeatedly under different source locations and speeds of sound. Results show that this iteration always leads to a convergent range of source locations and speeds of sound, from which approximate values of source locations and speed of sound can be determined. Once this is completed, the signals from individual sources are separated by de-convoluting the mixed signals with respect to individual source locations. This blind source separation methodology is then applied to the heart sounds measured on volunteers to separate the aortic, pulmonic, mitral, and tricuspid sounds.

Session 4aUW

Underwater Acoustics: Three-Dimensional Underwater Acoustics Models and Experiments I

Ying-Tsong Lin, Cochair
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Marcia J. Isakson, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78713

Chair's Introduction—8:00

Invited Papers

8:05

4aUW1. Three-dimensional propagation in the open ocean: Observations and modeling. Kevin D. Heaney and Richard L. Campbell (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

A primary feature of ocean acoustic propagation is that vertical environmental scales and gradients are generally an order of magnitude smaller than horizontal scales. As such, ocean acoustic experiments and modeling have developed with the view that the in-plane propagation is primary. There are environments, however, where the horizontal variability can be significant enough to lead to a change in the observed acoustic field due to out-of-plane refraction or diffraction. This is particularly evident at very low frequencies, where the open ocean can no longer be considered deep water and the seafloor topography is not negligible. In this paper, several recent observations of the impact of three-dimensional propagation on global propagation will be presented, as well as a fully 3-D parabolic equation, which is efficient enough to handle very long range propagation at low frequencies. The final modeling illustration will be the impact of ocean mesoscale on arrival angle of low frequency long range signals. This is implications to the detection capability of systems like the Comprehensive Testban Treaty Organization.

8:25

4aUW2. Bottom-diffracted surface-reflected arrivals in the Philippine Sea. Ralph A. Stephen (Geol. & Geophys. MS 24, WHOI, 360 Woods Hole Rd., Woods Hole, MA 02543-1592, rstephen@whoi.edu), Peter F. Worcester (Scripps Inst. of Oceanogr., La Jolla, CA), Richard L. Campbell (OASIS, Inc., Lexington, MA), Ilya A. Udovydchenkov (Mitre Corp., Bedford, MA), and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

Bottom-diffracted surface-reflected (BDSR) arrivals were first identified in the 2004 Long-range Ocean Acoustic Propagation Experiment (LOAPEX) in the North Pacific (Stephen *et al.*, 2013, JASA, **134**, 3307–3317). For sources at long ranges in deep water, they can be observed throughout the water column, but at depths above the conjugate depth they are obscured by ambient noise and energy propagating in the ocean sound channel (PE predicted arrivals). In deep water (deeper than the conjugate depth), ambient noise and PE predicted arrivals are sufficiently quiet that BDSR paths, scattered from small seamounts, can be the largest amplitude arrivals observed. In the Ocean Bottom Seismometer Augmentation in the Philippine Sea (OBSAPS) Experiment in April–May 2011, we observed BDSR arrivals on ocean bottom seismometers at relatively short ranges (less than 50 km). The experiment was designed to further define the characteristics of the BDSRs and to understand the conditions under which BDSRs are excited and propagate. BDSR arrivals are distinct, with little indication of coda or reverberation, and the scattering point appears to be discrete. The BDSR mechanism provides a means for acoustic signals and noise from distant sources to appear with significant strength on the deep seafloor. [Work supported by ONR.]

8:45

4aUW3. The role of water column variability on three-dimensional sound propagation in shallow water. Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

Simultaneous measurements of sound field and the sound speed profile perturbed by passage of internal waves in shallow water and the resulting modal evolution in space-time domain is reported. The waveguide modal behavior for broadband signals centered at 100, 200, and 300 Hz with the bandwidth of 60 Hz during the Shallow Water 2006 (SW06) experiment is studied using an L-shaped hydrophone array. Several cases are considered where the angle of the internal wave front with the acoustic track ranges from −8 to + 83 degrees. Mode filtering obtained from the vertical array and tracking the filtered mode along the horizontal array reveals that individual modes propagating between the source and receiver arrive at different angles. Using the method of vertical modes and horizontal rays the phase of the acoustic field is calculated and the change in the angle of arriving ray at the L-shaped array, that is the horizontal angle of refraction, is obtained. A 3D PE model is used for comparison between the calculations and the experimental data. [Work supported by ONR 322OA.]

Contributed Papers

9:05

4aUW4. Bathymetric refraction effects associated with marine pile driving. David Dall'Osto (Appl. Phys. Lab. at Univ. of Washington, Seattle, WA) and Peter H. Dahl (Appl. Phys. Lab. and Dept. of Mechanical Eng., Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dahl@apl.washington.edu)

Measurements of underwater noise from vibratory pile driving from a marine construction site in Puget Sound, Washington, are studied using line array-based measurements made at range 16 m from the pile source and single hydrophones at range 417-m on one transect, and at ranges 207-m and 436-m on another transect running approximately parallel to a sloping shoreline. Using adiabatic modes, the field from an incoherent line source model is propagated along these two transects. This approach works well except for the 436-m measurements located on the along-shore transect, where the observed levels for frequencies <300 Hz are significantly lower than predicted. These observations are interpreted in the context of shadow zones associated with bathymetric refraction for a downward sloping beach [Deane and Buckingham, J. Acoust. Soc. Am. **93**, 1319 (1993)]. The modal amplitude due to horizontal refraction is evaluated with the parabolic wave equation, written in terms of the depth dependent phase velocity of each mode [Ballard, Proc. Meetings Acoust.ics **15**, 070001 (2012)]. The results are significant as sloping beaches form many of the boundaries to waters where pile driving occurs, and where protective environment underwater noise monitoring is necessary, such as in Puget Sound.

9:20

4aUW5. Matched mode geoacoustic inversion of broadband signals in shallow water. Hefeng Dong (Dept. Electronics and Telecommunications, NTNU, Norwegian Univ. of Sci. and Technol., Trondheim NO 7491, Norway, hefeng@iet.ntnu.no), Mohsen Badiey (Univ. of Delaware, Newark, DE), and Ross Chapman (Univ. of Victoria, Victoria, British Columbia, Canada)

A matched mode geoacoustic inversion is presented for broadband acoustic signals recorded in experiments at a shallow water region of the New England Bight. First the modal behavior of the waveguide in the presence of 3D effects due to the water column and bottom bathymetry is examined. Spatial measurements of the temperature and salinity profiles and bathymetry provide data for calculating 3D propagation of the broadband acoustic wave field. Frequency shift of the broadband signal along one of the source-receiver paths in the experiment is attributed to internal waves propagating in the water column. Geoacoustic model parameters of the seabed along a source-receiver track with constant bathymetry are estimated by matched mode processing through matching the phase of the mode signals recorded at a Vertical Line Array. A warping transform was applied to identify and isolate modes, and the filtered modes were inverse transformed back to the time domain to reconstruct the mode signals. The inversion results are compared with the results from other inversion methods to assess the performance of the inversion scheme for estimating reliable values of the geoacoustic parameters in the experiment region.

9:35

4aUW6. Horizontal coherence of sound propagation in the presence of internal waves on the New Jersey continental shelf. Lin Wan and Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., 104 Robinson Hall, Newark, DE 19716, wan@udel.edu)

The knowledge of spatial coherence of sound field is required for sonar application and array design. In shallow water waveguides, the spatial and temporal evolution of the temperature field induced by internal waves will cause signal fluctuations and variations in spatial coherence. During the Shallow Water Acoustic Experiment 2006, the three dimensional temperature field of internal waves was measured and reconstructed while acoustic signals were simultaneously transmitted between various sources and an L-array. These internal wave events have been used to investigate the internal wave effect on temporal coherence and vertical structures of acoustic normal modes [Wan and Badiey, J. Acoust. Soc. Am. **135**, 2014]. In this paper, the horizontal structures of acoustic normal modes are examined. The arrival angles of normal modes are analyzed. The horizontal coherence of sound field is obtained at different frequencies for these internal wave events. The relationship between horizontal coherence and internal wave parameters, such as propagation direction, speed, and amplitude, is discussed. [Work supported by ONR322OA.]

9:50–10:10 Break

Invited Papers

10:10

4aUW7. Laboratory scale measurements of three-dimensional sound propagation in the presence of a tilted penetrable bottom. Frédéric Sturm (Ctr. Acoustique, LMFA UMR CNRS 5509, Ctr. Acoustique, Ecole Centrale de Lyon, 36, Ave. Guy de Collongue, Ecully 69134, France, frederic.sturm@ec-lyon.fr), Jean-Pierre Sessarego (LMA, UPR CNRS 7051, Marseille, France), and Alexios Korakas (Ctr. Acoustique, LMFA UMR CNRS 5509, Ecully Cedex, France)

Results of laboratory-scale measurements of long-range across-slope pulse propagation in three-dimensional wedge-shaped oceanic waveguides with a sandy bottom are reviewed. The experimental data considered were collected during two laboratory-scale experimental campaigns that were led in 2006 and 2007 (after a required calibration phase) in the large indoor shallow-water tank of the LMA-CNRS laboratory in Marseille. Operational frequencies and water depths were chosen to produce a reduced number of propagating modes so as to facilitate the analysis of the received signals. The experimental data contain strong 3-D effects like mode shadow zones and multiple mode arrivals, all consistent with well-known three-dimensional effects described in the literature. The data are compared with numerical solutions obtained using a fully three-dimensional parabolic equation based model. Comparisons are performed in both time domain (received signals at distinct ranges and depths) and frequency domain (TL-*versus*-range curves at distinct depths for several discrete frequencies). In both cases, the experimental data are in good agreement with the numerical predictions.

10:30

4aUW8. Measurements of three-dimensional acoustic propagation in a scale-model canyon. Jason D. Sagers (Environ. Sci. Lab., Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu)

Scale-model acoustic propagation experiments were conducted in a laboratory tank to investigate three-dimensional (3D) propagation effects induced by range-dependent bathymetry. The model bathymetry, patterned after measured bathymetric data, represents a portion of the Hudson Canyon at 1:7500 scale and was fabricated from closed-cell polyurethane foam using a computer-numerically controlled (CNC) milling machine. In the measurement apparatus, a computer-controlled positioning system locates the receiving hydrophone at user-defined points in 3D space while the stationary source hydrophone emits broadband pulsed waveforms. Precise control of the receiving hydrophone permits the creation of synthetic arrays from which horizontal and vertical beamforming is performed. Results are shown for propagation paths along and across the axis of the canyon where the received time series are post-processed to estimate travel times, transmission loss, and horizontal and vertical arrival angles. [Work supported by ONR.]

10:50

4aUW9. Laboratory measurements of sound propagation in a continuously stratified ocean containing internal waves. Likun Zhang, Harry L. Swinney (Ctr. for Nonlinear Dynam., Univ. of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199, lzhang@chaos.utexas.edu), and Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

A tank experiment is being conducted to explore sound propagation in a continuously stratified ocean containing internal gravity waves that produce fluctuations in acoustic signals. The tank is filled with water of increasing salinity with increasing depth in the fluid; the variation in salinity results in a 150 m/s variation of the sound speed over the 50 cm-depth fluid. Sound arrival times determined in three-dimensional hydrophone scans map out the sound diffraction pattern in the stratified fluid. Good agreement is found between the measurements and results obtained from wavefront modeling by ray acoustics. Sound fluctuations produced by internal waves are measured for several acoustic track locations and at different phases of the internal waves that radiate from a mechanical wave generator. Internal wave fields measured by particle image velocimetry provide input for a parabolic equation model, which is used to predict the sound fluctuations for comparison with measurements. This research is designed to improve the understanding and modeling of sound propagation and scattering by internal waves in the realistic oceans. [Research support: The 2013-14 ASA F. V. Hunt Postdoctoral Research Fellowship (L. Zhang) and ONR MURI Grant N000141110701 (WHOI).]

Contributed Papers

11:10

4aUW10. Comparison of measured acoustic reflection fluctuations and estimates based on roughness. Nicholas P. Chotiros and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas at Austin, PO Box 8029, Austin, TX 8713-8029, chotiros@arlut.utexas.edu)

The extent to which the roughness of the seabed can account for the acoustic reflection fluctuations is investigated. The roughness of selected areas of the seabed off Panama City, FL, was measured with a laser profiler, as part of Target and Reverberation EXperiment of 2013 (TREX13). The site may be characterized as having small-scale roughness due to bioturbation overlying larger sand ripples due to current activity. It was largely composed of sand with shell hash crossed by ribbons of softer sediment at regular intervals. Simultaneous with the roughness measurement, the seabed reflection was measured with a wide beam echo sounder centered about 10 kHz. Both instruments were deployed on a remotely operated vehicle (ROV) at a height of approximately 2 m above the seabed. Significant fluctuations in the acoustic reflection were observed. By modeling the acoustic scattering due to the roughness, the extent to which the roughness can account for the acoustic fluctuations will be investigated. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

11:25

4aUW11. Influence of short time scale water column fluctuations on broadband signal intensity and beam spreading. Justin Eickmeier and Mohsen Badiey (College of Earth, Ocean and Environment, Univ. of Delaware, Robinson Hall Rm. 112B, Newark, DE 19716, jeickmei@udel.edu)

The environment in a recent experiment exhibited short time scale isotherm depressions and elevations in the temperature profile of the water column. This dynamic behavior is significantly pronounced over a 2 hour period (between 70–90 m depth) during a 24 hour deployment. High frequency broadband transmissions (22–28 kHz) were sent between a stationary source (5 m above the seafloor) and an 8-element vertical hydrophone array (4.5 m above the seafloor) in an approximate depth of 100 m with 1 km separation. Vertical beamforming of measured impulse response across all array elements and application of Gaussian steering revealed strong correlation between vertical temperature profiles and angular spread of the direct path receptions. Inherently a 3D problem, we consider a 2D approach to show beam fluctuations as a function of the environment. 2D PE modeling is driven by measured sound speed profiles to calculate the acoustic field between source and receiver and to beamform across an ideal vertical array for data/model comparison. Over time, fluctuations in the intensity of the acoustic beam, spatial path and angular spread of the direct path signal can be attributed to the vertical oscillations of isotherms in the water column. [Work supported by ONR321.]

11:40

4aUW12. An assessment of the effective density fluid model for scattering from poroelastic sediments with inclusions. Anthony L. Bonomo, Nicholas P. Chotiros, and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

The effective density fluid model (EDFM) was developed to approximate the behavior of sediments governed by Biot's theory of poroelasticity. Previously, it has been shown that the EDFM predicts reflection coefficients and backscattering strengths that are in close agreement with those of the full Biot model for the case of a homogeneous poroelastic half-space. However, it has not yet been determined to what extent the EDFM can be used in place of the full Biot model for other cases. In this work, the finite element method is used to compare the scattered pressure predicted using the EDFM with the predictions of the full Biot model for the case of a poroelastic 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.]

Session 4pAA

Architectural Acoustics: Classroom and Other Room Acoustics

Richard J. Ruhala, Chair
Mechanical Engineering, Kennesaw State University, 1100 South Marietta Parkway, KSU - SPSU - ME Department, Marietta, GA 30060

Contributed Papers

1:15

4pAA1. Acoustical design of schools on military bases: Meeting the needs of DoDEA's 21st century school Initiative and LEED acoustical requirements. Jessica S. Clements (Newcomb & Boyd, 303 Peachtree Ctr. Ave. NE, Ste. 525, Atlanta, GA 30303, jclements@newcomb-boyd.com)

In their efforts to improve student learning environments, DoDEA (Department of Defense Education Activities) has included acoustical requirements in their 21st Century Schools initiative design guidelines. Recently, DoDEA projects have been required to also achieve LEED certification. The acoustical requirements for LEED schools outlined in IEQ prerequisite 3 (IEQp3) Minimum Acoustical Performance are referenced in part from ANSI S12.60 Acoustic Performance and Guidelines for Schools. Ideally the schools would achieve both the LEED acoustic prerequisite and the additional acoustic credit in order to achieve most of the goals of ANSI S12.60. However, meeting some aspects of the 21st Century Schools design guideline seems to preclude achieving the additional Acoustical Performance credit. Each of these guidelines includes different aspects of architectural acoustics design requirements. This leads to potentially three different design guidelines affecting a single project. The presentation will compare and contrast the design requirements of ANSI S12.60-2010 Part 1, LEEDv4 Minimum Acoustic Performance IEQp3, LEEDv4 Acoustic Performance Credit IEQc9, and the DoDEA 21st Century Design Guideline. Discussion will include examples from recent projects and resolutions used between conflicting design requirements.

1:30

4pAA2. An investigation into the acoustics of different sized open plan and enclosed Kindergarten classrooms. Kiri T. Mealings (Linguist, Macquarie Univ., Level 3 AHH Macquarie University, Sydney, New South Wales 2936, Australia, kiri.mealings@students.mq.edu.au), Jörg M. Buchholz (National Acoust. Labs., Sydney, New South Wales, Australia), Katherine Demuth (Linguist, Macquarie Univ., Sydney, New South Wales, Australia), and Harvey Dillon (National Acoust. Labs., Sydney, New South Wales, Australia)

Open plan classrooms, where several class bases share the same space, have recently re-emerged in Australian primary schools. This study compared the acoustics of four different Kindergarten classrooms: an enclosed classroom with 25 students, a double classroom with 44 students, a linear fully open plan triple classroom with 91 students, and a semi-open plan K-6 classroom with 205 students. Ambient noise levels, intrusive noise levels, occupied background noise levels, and teacher's speech levels were recorded during different activities. Room impulse responses using logarithmic sweeps were also recorded for different teaching scenarios. From these recordings, signal-to-noise ratios, speech transmission index scores, and reverberation times were calculated. The results revealed much higher intrusive noise levels in the two largest open plan classrooms, resulting in signal-to-noise ratios and speech transmission index scores to be well below those recommended in classrooms with students of this age. Additionally, occupied background noise levels in all classrooms were well above recommended levels. These results suggest students may have difficulty listening and learning in open plan classrooms and teachers are likely to strain their voice. Therefore, open plan classrooms are unlikely to be appropriate learning environments for young children unless high intrusive noise levels are moderated.

1:45

4pAA3. Sound levels and speech privacy in an open space academic learning environment. Richard J. Ruhala, Aaron Baker, and Ermal Shpuza (Kennesaw State Univ., 1100 South Marietta Parkway, KSU - SPSU - ME Dept., Marietta, GA 30060, rruhala@spsu.edu)

Undesired sound and speech privacy in a large educational space or open office environment are persistent problems for occupants. The architectural design studio on the campus of Kennesaw State University (formerly Southern Polytechnic State University) is used as a case study. Students have desk space very similar to those in professional open offices, and are used for individual and group projects. Simultaneously classes are held where instructors or students will be presenting to their class in an area that is only partially partitioned off from the rest of the design studio. For the first set of *in situ* experiments, sound levels are recorded over several days at various locations and correlated with learning activity. Second, broadband noise is used to measure the reduction of spectrum levels between source and receiver, at different locations. Finally, students and professors participate in an anonymous questionnaire regarding their perceptions of the acoustic quality of the space; specifically regarding the speech intelligibility of professors, level of noise distractions, and preferences of when to work in the studio. This study aims to understand the acoustic impact of human activity occurring concurrently in a large open academic environment.

2:00

4pAA4. Classroom acoustics for vocal health of elementary school teachers. Jennifer K. Whiting, Zachary R. Jensen, Mark L. Berardi, Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lundjenny@comcast.net), and Eric J. Hunter (Dept. of Commun. Sci. and Disord., Michigan State Univ., East Lansing, MI)

School teachers have an elevated risk of voice problems due to vocal demands in the workplace. ANSI SI2.60-2002 provides a standard for classroom acoustics, but it focuses primarily on students and unoccupied classroom settings. This presentation explores a preliminary study of six elementary school teachers that included measurements of architectural acoustics parameters and noise-levels of their classrooms, as well as their speech levels and fundamental frequencies over the course of a school day. The measurement methods and speech trends are discussed for the various cases, demonstrating that classroom acoustics standards may benefit from greater attention to teacher vocal health.

2:15

4pAA5. Identifying vocal changes due to environmental changes: Real-time Lombard effect. Zachary R. Jensen, Jennifer K. Whiting, Timothy W. Leishman (Acoust. Res. Group, Phys. and Astronomy Dept., Brigham Young Univ., Provo, UT 84602, zjens1@gmail.com), and Eric J. Hunter (Commun. Sci. & Disord., Michigan State Univ., East Lansing, MI)

School teachers are known to have an elevated risk of voice problems due to speaking demands in the workplace, as well as noise conditions in the classroom. Six elementary school teachers were recorded for a full school day using a neck-worn accelerometer and microphone. Classroom noise levels were also recorded using seven stationary microphones distributed randomly throughout the room. A method was developed that used the accelerometer signal to identify when a teacher is speaking, subsequently calculating a speech-to-noise ratio from six hours of teaching. Speech levels and fundamental frequencies were assessed through the accelerometer signal. The former were compared to temporally and spatially proximate noise levels from the neck-worn microphone when the teacher was not speaking. The speech data was also compared to spatially averaged and temporally proximate classroom noise levels when the teacher was not speaking. While unique in the duration of the observation for the various teachers, the speech-to-noise ratio exhibited trends related to those suggested in previous studies and is congruent with the Lombard effect. A good understanding of how a teacher compensates for realistic occupational noise conditions can lead to better recommendations for the teacher to retain good vocal health. The method and speech trends will be discussed, and future methodological improvements will be suggested.

2:30

4pAA6. Beyond Lombard: Quantitative changes due to noise and reverberation. Nathan G. Eyring, Mark L. Berardi, Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., N-283 ESC,Provo, UT 84602, eyring.acoustics@gmail.com), and Eric J. Hunter (Dept. of Commun. Sci. and Disord., Michigan State Univ., East Lansing, MI)

Forty-five subjects performed a short vocal task in two different rooms: a variable-acoustics room and an anechoic chamber. The subjects were taken back and forth between the two rooms using a deception protocol. Each time they entered the variable-acoustics room, its acoustical characteristics had been changed without a change in visual appearance. The changing characteristics involved two background noise conditions and two reverberation conditions. Subjects responded to questions about their comfort and perception of the environmental changes. Analysis was performed on the second and third sentences of the rainbow passage. Objective acoustical metrics and perceptual responses were compared for the different settings. In contrast to males, females raised their fundamental frequency (F0) in concert with their raised vocal levels in response to changes in both loudness and reverberation. A high correlation existed between pitch strength and F0. Factor analysis also revealed that F0 and vocal level were more correlated in females than males.

2:45

4pAA7. Acuity to noise and occupational voice risks in simulated acoustical environments. Mark L. Berardi, Jennifer K. Whiting, Nathan G. Eyring, Michael K. Rollins, Zachary R. Jensen (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ.,N283 ESC, Provo, UT 84602, markberardi12@gmail.com), Eric J. Hunter (Dept. of Commun. Sci. and Disord., Michigan State Univ., East Lansing, Utah), and Timothy W. Leishman (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

School teachers are known to have an elevated risk of voice problems due to the vocal demands in their work environments. Forty-five participants (20 females, 25 males, 7 elementary school teachers, and 38 college-age adults) performed a short vocal task in two different rooms: a variable-acoustics room and an anechoic chamber. The subjects were taken back and forth between the two rooms using a deception protocol. Each time they entered the variable-acoustics room, the acoustical characteristics (two background noise conditions and two reverberation conditions) had been changed without a visual appearance of change. Analysis was conducted on recorded second and third sentences of the first paragraph of the Rainbow Passage. Results revealed that differences in response to reverberation was gender specific. Additionally, school teachers seemed to be more susceptible to the noise condition.

3:00–3:15 Break

3:15

4pAA8. Low-cost redesign of a music rehearsal space for high school ensembles. Gabe Murray (Mech. Eng. Dept., Iowa State Univ., 2025 Black Eng., Ames, IA 50011, glmurray@iastate.edu) and Thomas Burns (Starkey Hearing Technologies, Eden Prairie, MN)

This presentation describes the redesign of an existing multi-purpose practice room used by high school concert bands, drumlines, and choral groups. Resource and cost restrictions limited the redesign and evaluation process for surface finishes only. The existing space had minimal early reflections and reverberation due to copious amounts of thick acoustic wall panels, resulting in a dead sound for choral and band use. The biggest challenge was to reposition the wall panels by distributing them in a checkerboard pattern, thereby balancing them with the reflective surfaces of the painted cinderblock underneath. In addition, one third of the absorptive suspended ceiling tile was replaced with 5/8” gypsum panels, and one third of the ceiling tile was removed altogether in order to couple the interstitial space above the suspended ceiling and increase the effective volume of the space. Odeon was used to create auralizations of various configurations by convolving anechoic recordings of a snare drum with the computed impulse responses, which will be demonstrated. This work was done as part of the Eagan, MN, High School Mentor Program and a professional partnership with Starkey Hearing Technologies.

3:30

4pAA9. The perceptual influence of the cabin acoustics on the reproduced sound of a car audio system. Neofytos Kaplanis, Søren Bech (Bang & Olufsen, Peter Bangs Vej, 15, Struer 7600, Denmark, neo@bang-olufsen.dk), Sakari Tervo, Jukka Pätynen, Tapio Lokki (Dept. of Media Technol., Aalto Univ., Espo, Finland), Toon v. Waterschoot (Elec. Eng., ESAT-STADIUS/ETC, KU Leuven, Leuven, Belgium), and Søren H. Jensen (Electron. Systems, Aalborg Univ., Aalborg, Denmark)

A significant element of audio evaluation experiments is the availability of verbal descriptors that can accurately characterize the perceived auditory events. In terms of room acoustics, understanding the perceptual effects of the physical properties of the space would enable a better understanding of its acoustical qualities, and stipulate perceptually relevant ways to compensate for the subsequent degradations. In contrast to concert halls, perceptual evaluation of everyday-sized and less reverberant spaces has been a challenging task, and literature on the subject is limited. In this study, a sensory evaluation methodology [Lokki *et al.*, J. Acoust. Soc. Am. **132,** 3148–2161 (2012)] was employed to identify the most relevant attributes that characterize the influence of the physical properties of a car cabin on the reproduced sound field. A series of *in-situ* measurements of a high-end car audio system was performed for different physical settings of the car’s cabin. A novel spatial auralization methodology was then used, and participants were asked to describe verbally the perceived acoustical characteristics of the stimuli. The elicited attributes were then analyzed following a previous review [Kaplanis *et al.*, in *55th Int. Conf. Aud. Eng. Soc.* (2014)] and possible links to the acoustical properties of the car cabin are discussed. [This study is a part of Marie Curie Network on Dereverberation and Reverberation of Audio, Music, and Speech. EU-FP7 under agreement ITN-GA-2012-316969.]

3:45

4pAA10. Investigations on multi-slope energy decay formation within single-space monumental structures by diffusion equation modeling and intensity probe measurements. Zühre Sü Gül (MEZZO Studyo LTD, METU/ODTU Technopolis Galyum Blok No 21 - A, Cankaya, Ankara 06800, Turkey, zuhre@mezzostudyo.com), Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY), and Mehmet Čalışkan (Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

Multiple domed superstructures with different volumetric and material attributes are previously investigated using in-situ acoustical measurements and acoustical simulation methods for analyzing their contributions to multi-slope energy decay formation. Relevant acoustical parameters including decay rates and decay times are computed by applying the Bayesian decay analysis. Initial results indicate double- or triple-slope decay characteristics for specific measurement configurations for selected cases. The ongoing research is aimed at explaining the probable reasons of multiple decay formation as observed in such single space enclosures. Diffusion equation modeling (DEM) and intensity probe measurements are utilized to observe spatial sound energy distributions and energy flow vectors. Both computed and measured flow vectors indicate the contribution of central dome to the later energy feedback. For a specific case the trial by DEM over virtual model with floor with marble instead of carpet resulted in a centrally concentrated energy, as a result preventing the multi-slope formation. The results support the argument that non-diffuse sound fields due to geometrical and material characteristics of superstructures provide the circumstances for non-exponential energy decay formation.

4:00

4pAA11. Minimum phase acoustic systems: An intuitive model study.
Sahar Hashemgeloogerdi and Mark F. Bocko (Elec. and Comput. Eng., Univ. of Rochester, 405 Comput. Studies Bldg., Rochester, NY 14627, shashemg@UR.Rochester.edu)

The zeros and poles of minimum phase systems lie within the unit circle in the z-plane, which ensures the existence of a stable inverse. In acoustics, the question if a system is minimum phase has important consequences, such as the ability to find an inverse of a room's acoustic transfer function to compensate for reverberation. In this paper, we provide an intuitive model of minimum phase acoustic systems based upon physical arguments. Using the image method, we compute the transfer function of generic one, two, and three-dimensional acoustic enclosures with a sound source and a microphone placed at arbitrary locations. We find the relationship between the direct sound field and the reverberant sound field that ensures the minimum phase property of the system. Essentially, the system retains the minimum phase property if the radius of reverberation, i.e., the distance from the source where the direct sound and reverberant sound become equal in magnitude, lies outside of the boundaries of acoustic enclosure. We show that a one-dimensional acoustic system always is minimum phase for the entire range of source-microphone locations and wall reflection coefficients. However, two and three-dimensional enclosures display minimum phase response for limited ranges of the system parameters.

4:15

4pAA12. Evaluation of a three-way omnidirectional sound source for room impulse response measurements.
David A. Dick, Matthew T. Neal, Carol S. Tadros, and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dad325@psu.edu)

Commercially available omnidirectional loudspeakers, which are used to measure room impulse responses, typically have a limited bandwidth and are only omnidirectional below approximately 1kHz. To extend the operating bandwidth, a three-way omnidirectional source was built consisting of a subwoofer, a mid-frequency dodecahedron, and a high-frequency dodecahedron. The subwoofer contains two 10 in. drivers in a sealed box. The mid-frequency dodecahedron uses 12 4 in. mid-bass drivers. The high frequency dodecahedron was made with 12 closely spaced 3/4 in. dome tweeters, providing omnidirectional behavior up to approximately 5 kHz. The directivities of the dodecahedrons were validated by taking measurements in an anechoic chamber. The performance of the sources in a realistic setting was assessed using measurements taken in Eisenhower Auditorium, located at Penn State. Measurements were made with the three sources placed individually on the stage in the same location and compared to measurements made with the sources stacked one on top of the other. The stacked configuration yielded significantly different results than the individual configuration. Additionally, measurements were made rotating the individual sources on the stage to evaluate the effect of angular orientation on measured parameters. Parameter differences were found to be small up to the 4 kHz octave band. [Work supported by NSF Grant 1302741.]

4:30

4pAA13. St. Bartholomew's parish hall acoustic analysis.
Christopher L. Barnobi (CSTI Acoust., 16155 Park Row Ste. 150, Ste. 101, Houston, TX 77084, chris@cstiacoustics.com), Ana Jaramillo (Olson Sound Design, Medellin, Colombia), and Adam Young (CSTI Acoust., Houston, TX)

St. Bartholomew's Catholic Church in Katy, Texas, includes a fellowship hall with a great dome. Parishioners use the space for social events including meals and regular bingo games. High reverberation times in the fellowship hall reduced speech communication and acoustic comfort. Measurements were conducted to assess the reverberation time of the room before and after applying treatment to the space. Sabine reverberation time analysis was used to specify the recommended sound absorption for the fellowship hall. EASE analysis of the space reveals more interesting characteristics.

4:45

4pAA14. What is the necessary level of detail for an acoustic model?
Bruce C. Olson and Ana M. Jaramillo (Olson Sound Design, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu)

In order to evaluate the level of architectural detail needed in an acoustic model, several projects were selected. Using the software EASE, the authors created several models for each project and found that a high level of architectural detail was unnecessary to some extent in an acoustic model, which gained an advantage in modeling and calculation times.

Session 4pAB

Animal Bioacoustics: Audio Playback in Animal Bioacoustics

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

Chair's Introduction—2:00

Invited Paper

2:05

4pAB1. Multimodal communication in wolf spiders: Playback studies with visual and vibratory signals. George W. Uetz (Biological Sci., Univ. of Cincinnati, ML 0006, Cincinnati, OH 45221-0006, george.uetz@uc.edu), David L. Clark (Biology, Alma College, Alma, MI), Brent Stoffer, Elizabeth C. Kozak, Madeline Lallo (Biological Sci., Univ. of Cincinnati, Cincinnati, OH), and Heather Kane (Biology, Alma College, Alma, MI)

Previous studies of *Schizocosa ocreata* wolf spiders have shown that visual and vibration signals are equally capable of eliciting female receptivity, but that multimodal cues enhance female response. Studies also indicate that signals in both modes convey male quality information; females choose males with larger foreleg tufts or greater amplitude vibratory signals. Male spiders eavesdrop on both the visual and vibration signals of other males and exploit them to find mates. We examined female mate preferences and male eavesdropping using video-vibratory playback experiments. Female *S. ocreata* respond to playback of male courtship, showing responses to both visual and vibratory signals alone, as well as multimodal playback. Females also showed preference for male quality indicators in both sensory modes. Eavesdropping males responded to playback of male courtship with displays of their own, but responded to signal modes differently. Males displayed a higher rate of courtship tapping in response to isolated vibration signals compared to visual or multimodal stimuli. However, in choice tests, males responded with higher display rates with multimodal and visual stimuli. Results suggest that visual, vibratory and multimodal courtship signals provide information about potential mates and rivals, and that responses of males and females may differ depending on context.

Contributed Paper

2:25

4pAB2. Quantifying response in vocal behavior of fin whales to local shipping in the Southern California. John E. Joseph and Tetyana Margolina (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jejoseph@nps.edu)

Analysis of a year-long passive acoustic dataset collected at Thirtymile Bank from December 2012 to December 2013 revealed two main types of response, if any, in the vocalization behavior of fin whales to passing ships. The whales ceased calling or reduced the vocalization intensity and then resumed it after the ship passes, or the vocalization intensity increased during the passage and remained elevated for a short period of time afterward. While the response has been primarily observed in 20 Hz calls, several cases have been observed when the call pattern changed with 40 Hz calls recorded after the ship passed. Our study quantifies observed response in terms of both vocalization intensity and call rates. The latter is made possible by applying an image-based detection algorithm allowing for extraction of individual calls. Intensity variations and call rate changes related to ship sounds are statistically estimated, as well as the type and strength of the response to ship sound amplitude, frequency content and relative position of the whales to the passing ships. The received level of the ship sounds presumably heard by whales is modeled using an acoustic propagation model and a Monte Carlo approach to estimate the unknown whale location.

Invited Paper

2:40

4pAB3. The potential for acoustic communication in the "purring" wolf spider, *Gladicosa gulosa*. Alexander L. Sweger and George W. Uetz (Biological Sci., Univ. of Cincinnati, P.O. Box 210006, Cincinnati, OH 45221, swegera@gmail.com)

Vibration is an important part of the sensory world in spiders, and many species have adapted vibration as a major part of their conspecific communication. While nearly all male wolf spiders produce vibrations during courtship, the "purring" wolf spider, *Gladicosa gulosa,* also produces an acoustic signal in conjunction with its vibratory display. However, with limited previous research on this species, the evolutionary significance of this component remains unknown. Given that spiders are not known to possess sensory structures

for directly perceiving airborne sound, this raises a number of questions about the production, reception, and possible role of the signal. We measured male signal production and male/female responses to isolated acoustic signals on both vibrating (paper) and non-vibrating (granite) substrates. We found that male signals, both vibratory and acoustic, are only present in vibrating substrates. We also found significant differences in phonotaxis based on sex of the focal individual, stimulus type, and substrate type. These results suggest that the substrate plays an important role in both production and reception of the acoustic signal, and that under certain conditions, acoustic signaling may have a role in the communication network in this species.

Contributed Paper

3:00

4pAB4. Playback experiments for *Thryothorus ludovicianus* in urban backyard experiments. David P. Knobles (KSA, PO Box 8029, Austin, TX 78713, knobles@arlut.utexas.edu) and Mohsen Badiey (Univ. of Delaware, Newark, DE)

Acoustic playback experiments were performed from June 2014 to April 2015 for *Thryothorus ludovicianus* (Carolina Wren) in a backyard environment in Austin, Texas. Acoustic recordings were made for pre-playback, playback, and post-playback periods for multiple song types. The songs were obtained from recordings made in the backyard and surrounding neighborhood. Recordings of both the playback and acoustic responses of the bird subjects are made in coincidence with visual observations of the bird's behavior. Bird behavior is quantified in terms of the statistics of closest distance of approach, latency to approach, latency to sing, number of post-playback songs, number of flybys, percent of male-female duets, temporal correlation of individual song phrases, and number of singing competitions with neighboring subjects. Statistics of the properties of the measured spectrograms and spatial correlation properties for experiments are presented, and attempts are made to interpret the observed behavior and acoustical responses of the birds in the form of communication and territorial defense with the bird's ability to estimate distance using the degradation (transmission loss and reverberation) of the playback calls. It was also observed that *Thryothorus ludovicianus* will respond to playback calls of *Cardinalis cardinalis* (Northern Cardinal) and this interaction between bird species is examined.

Invited Paper

3:15

4pAB5. The use of acoustic playbacks with captive cetaceans. Jason N. Bruck (Dept. of Integrative Biology, Oklahoma State Univ., 501 Life Sci. Way, Stillwater, OK 74075, jbruck@okstate.edu)

Playbacks are a powerful tool in studies of animal communication and cognition. They can provide insights into referentiality, social memory, anti-predator strategies, sexual selection, as well as a host of other topics. In particular, playbacks have been a useful method for understanding the nature of wild dolphin communication using temporarily restrained animals. Unfortunately, there are a series of questions where complete social histories of subjects or certain controls are needed, but unavailable in wild study populations. Captive, free-swimming animals present a complementary set of subjects where one can address questions involving referentiality or dolphin social memory. Methods for conducting playbacks with captive groups differ greatly than for wild subjects. I will explore the procedural differences of these experiments, including overcoming difficulties associated with recording and presenting sounds in an acoustically challenging environment as well as how to measure response behavior. Furthermore, I will discuss the potential of captive acoustic playbacks by highlighting data collected from such studies including evidence for dolphin multi-decade social memory, context dependent cue usage and acoustic kin recognition.

Session 4pBAa

Biomedical Acoustics: Ultrasound Contrast Agents: Molecular Imaging Applications

Tyrone M. Porter, Cochair
Boston University, 110 Cummington Mall, Boston, MA 02215

Jonathan A. Kopechek, Cochair
University of Pittsburgh, 200 Lothrop Street, Pittsburgh, PA 15213

Invited Papers

1:00

4pBAa1. Molecular imaging using ultrasound and microbubbles. Flordeliza S. Villanueva (Medicine, Univ. of Pittsburgh, 200 Lothrop St., Pittsburgh, PA 15213, villanuevafs@upmc.edu)

Increasing insight into the molecular mechanisms of disease has created a need to visualize sub-cellular events in clinical populations in order for treatment paradigms to advance. Molecular imaging has emerged as an important translational tool for understanding disease pathogenesis, for which various probes and detection methods have been developed. Ultrasonic detection of molecular markers of disease using acoustically active particles is one such *in vivo* modality. Molecular imaging with ultrasound utilizes intravascular delivery of gas-filled microspheres (microbubbles) targeted to bind to disease-specific epitopes expressed by vascular endothelial cells. Microbubble binding occurs by virtue of targeting ligands attached to the microbubble surface, which confer specific binding to the endothelial target. In the presence of ultrasound, the bound microbubbles emit an acoustic signal which can be detected as a persistent contrast effect during ultrasound imaging with non-linear based detection strategies. In pre-clinical models, ultrasound molecular imaging has been used to detect early atherosclerotic disease, myocardial ischemic memory (recent myocardial ischemia), acute heart transplant rejection, angiogenesis, and for *in vivo* tracking of therapeutically administered stem cells. Such applications raise exciting possibilities for meaningful bedside translation of this imaging platform to address important clinical issues in an era of molecular medicine.

1:30

4pBAa2. Novel polymer-lipid assemblies for stimulus-responsive imaging contrast agents. Andrew P. Goodwin (Chemical and Biological Eng., Univ. of Colorado Boulder, 3415 Colorado Ave., 596 UCB, Boulder, CO 80303, andrew.goodwin@colorado.edu)

Stimulus-responsive macromolecular structures are of tremendous importance for the next generation of tools for *in vivo* imaging and site-directed therapy. In this talk, I will present our research group's work regarding the creation of self-assembled amphiphilic structures that respond to the presence of biomarkers that present as hallmarks for disease such as deep venous thrombosis and cancer. In one instance, the mechanical properties of synthetic, lipid-shelled microbubbles could be manipulated to sense for the biomarker thrombin using DNA hybridization. These microbubbles were found to have "on-off" ratios of ~100 and could highlight regions of elevated thrombin in real time in a live rabbit model. In another research direction, low-contrast nanoemulsions were designed to vastly increase their detection capabilities in response to chemically-induced changes in their surface properties. Both examples showcase a method for obtaining site- or biomarker-specific contrast enhancement for deep tissue imaging using commercially available technologies and have excellent promise for advancement to clinical use.

2:00

4pBAa3. Molecular imaging with ultrasound: Steps toward clinical translation. Juergen K. Willmann (Radiology, Stanford, 300, Pasteur Dr., Rm. H1307, Stanford, CA 94305-5621, willmann@stanford.edu)

Molecular imaging provides the ability to non-invasively assess expression levels of molecules by measuring imaging signals generated with the help of molecularly targeted contrast agents. These contrast agents can be directed to bind various molecular targets, thereby visualizing and quantifying disease processes at the molecular level *in vivo*. In recent years, the field of molecular imaging has been rapidly expanding and involves multiple imaging modalities. Among those, ultrasound imaging is a widely available, relatively inexpensive, and real-time imaging technique without exposing patients to radiation. It is already the first-line radiological imaging modality for many organs and disease processes in patients. By combining the these advantages of ultrasound with the ability to image molecular signatures using novel molecularly targeted ultrasound contrast agents, molecular ultrasound imaging is a quickly evolving imaging strategy that has great potential as a highly accurate and quantitative method for various clinical applications, including earlier detection of for example cancer, characterization of focal lesions, and quantitative monitoring of various disease processes at the molecular level. The objective of this presentation is to provide an overview on the principles of ultrasound molecular imaging and to highlight current translational studies to move ultrasound molecular imaging approaches into first in human clinical applications.

Contributed Papers

2:30

4pBAa4. Laser-induced vaporization and acoustic-induced cavitation of droplets. Jaesok Yu (Dept. of BioEng., Univ. of Pittsburgh, 3550 Terrace St., Scaife Hall 958, Pittsburgh, PA 15261, JAY49@pitt.edu), Man M. Nguyen, and Kang Kim (Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

In this study, micro-droplets developed for photoacoustic imaging and drug delivery have been evaluated *in vitro*. Induced by a short laser pulse, perfluoropentane mixed with optical dye was vaporized and expanded up to about 20 times of initial diameter of 3–10 micron, generating strong broadband photoacoustic signals. It was found these vaporized droplets became less stable and prone to rupture to ultrasound pulses. The broadband inertial cavitation signals by a very short ultrasound pulse can provide high spatial resolution in passive cavitation imaging. For feasibility test, passive cavitation imaging algorithms were implemented in a commercial ultrasound open platform with a linear array transducer, centered at 5 MHz. The system was synchronized with a 1 MHz unfocused single element transducer. 700kPa single-cycle excitation ultrasound pulse was induced to a 580 μm inner diameter polyethylene tube containing micro-droplets. The cavitation imaging was performed before and after vaporizing droplets by laser. Broadband emissions of 3–7MHz were observed only with vaporized droplets. These preliminary results show the feasibility of cavitation imaging of vaporized droplets with a short ultrasound excitation pulse for improved spatial resolution and could lead to further *in-vivo experiments*.

2:45

4pBAa5. Numerical investigation of the nonlinear attenuation and dissipation of acoustic waves in a medium containing ultrasound contrast agents. Amin Jafari Sojahrood, Raffi Karshafian, and Michael C. Kolios (Dept. of Phys., Ryerson Univ., 350 Victoria St., Toronto, Ontario M5B2K3, Canada, amin.jafarisojahrood@ryerson.ca)

The presence of microbubble ultrasound contrast agents (UCAs) in path of an ultrasound beam increases the attenuation of the medium. A detailed knowledge about the medium attenuation in the presence of UCAs is critical for optimizing ultrasound applications. In this study, a model incorporating the nonlinear attenuation of microbubbles is developed by deriving the complex and real part of the wave number from the Calfish model. Results showed that when UCAs are sonicated by their linear resonance frequency (fr), the effective attenuation of the medium can potentially decrease as the pressure increases, in agreement with experimental observations of the attenuation of monodisperse microbubbles. When sonicated with their pressure dependent resonance frequency (fs), the effective attenuation of the medium is smaller than the case of sonication with fr, but only below a pressure threshold (Ps). Above Ps, the effective attenuation increases abruptly (e.g. ~3 fold) and becomes significantly more than the attenuation during sonication with fr. The attenuation in the subharmonic (SH) regimes of oscillations (sonication with n*fr where n = 2, 3.) are considerably smaller compared to the cases of sonication with fr and fs (~10 times). The attenuation undergoes a sharp increase concomitant with the generation of SH oscillations.

THURSDAY AFTERNOON, 21 MAY 2015 — KINGS 2, 3:15 P.M. TO 5:30 P.M.

Session 4pBAb

Biomedical Acoustics: Therapeutic Ultrasound and Bioeffects

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

Contributed Papers

3:15

4pBAb1. An equivalent source model for simulating high intensity focused ultrasound fields using a nonlinear parabolic equation. Vera A. Khokhlova, Petr V. Yuldashev, Pavel B. Rosnitskiy (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow 119991, Russian Federation, va.khokhlova@gmail.com), Wayne Kreider, Adam D. Maxwell, Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

The nonlinear parabolic Khokhlov-Zabolotskaya (KZ) equation is commonly used for simulation of nonlinear focused fields generated by medical ultrasound transducers. Axially symmetric codes can be easily implemented on personal computers with simulation times on the order of tens of minutes even for the most extensive simulations of shock formation conditions. However, model accuracy can be limited for high intensity focused ultrasound sources with large convergence angles. In previous studies, it was shown that nonlinear parabolic modeling can provide good matching with measurements and full diffraction modeling in the focal region for both single element piston sources and multi-element arrays. Boundary conditions for the modeling were adjusted by varying the aperture of an equivalent planar source with a uniform pressure magnitude and a phase distribution for parabolic focusing. With this approach, there were slight discrepancies in the lateral pressure distributions in the focal plane. Here, it is proposed that an additional variation in the position of the equivalent source enables matching both axial and lateral distributions of the real HIFU sources with higher accuracy. Several examples are presented to compare modeling and measurements of the fields generated by laboratory and clinical HIFU transducers. [Work supported by RSF 14-12-00974 and NIH EB007643.]

3:30

4pBAb2. A sum-of-harmonics time-domain method to distinguish harmonic and broadband signals during passive acoustic mapping of ultrasound therapies. Erasmia Lyka, Christian Coviello (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg., Headington, Oxford OX3 7DQ, United Kingdom, erasmia.lyka@eng.ox.ac.uk), Richard Kozick (Bucknell Univ., Lewisburg, PA), and Constantin C. Coussios (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Passive Acoustic Mapping (PAM), a novel technique for real-time monitoring of ultrasound-based therapies, performs passive beamforming of non-linear acoustic emissions simultaneously received on an array of transducers. These emissions can be classified according to their frequency content into harmonic, indicative of nonlinear scattering, stable cavitation or tissue boiling, and broadband, indicative of inertial cavitation. However, the magnitude of coherent reflections often greatly exceeds that of incoherent broadband emissions arising from inertial cavitation and the ability to distinguish these two components is, thus, key to successful detection and mapping of inertial cavitation. We propose a novel time-domain-based filter that uses a parametric model in order to estimate a time-varying amplitude harmonic component in the presence of a lower-amplitude broadband signal. Performance evaluation on simulated and experimental data has shown that the model is able to accurately detect a time-varying amplitude harmonic signal for harmonic-to-broadband amplitude ratio as high as 20 dB, even in short data lengths, decreasing the estimation error by at least 80% compared to conventional comb filtering, and it yields passive acoustic maps of better accuracy and spatial resolution. Sum-of-harmonics is expected to become a valuable component of real-time PAM and significantly contribute toward its clinical adoption.

3:45

4pBAb3. Temperature measurement using backscattered ultrasonic power for non-invasive thermometry during HIFU ablatio. Jeremy Chenot and David Melodelima (LabTAU - INSERM U1032, 151 cours Albert Thomas, Lyon 69003, France, David.Melodelima@inserm.fr)

The temperature dependence of ultrasonic tissue parameters has been reported extensively. In this work, the relationship between changes in ultrasound backscattered power and temperature during HIFU treatments was studied. An analytical model was developed based on attenuation, backscattered coefficient, velocity, and thermal expansion to predict temperature changes. *In vitro* and *in vivo* tests were conducted in liver to confirm simulations. HIFU treatments were performed using a focused transducer working at 3 MHz. The radius of curvature and the diameter were both 70 mm. An ultrasound imaging probe working at 7.5 MHz was placed in the center of the HIFU transducer. Long exposure time (120 s) was used to observe smooth temperature increase from 37 to 80 °C. Simulations predicted that backscattered power increased nearly logarithmically with temperature over the range from 37 °C to 80 °C. The model predicted a linear increase 15 dB. A linear increase 12 dB was measured in ultrasound backscattered power during experiments. The tissue temperature increase estimated using backscattered energy correlated well (r = 0.79) with temperature measurements performed using thermocouples. This linear relationship between changes in the backscattered energy and actual temperature was observed up to 80 °C. Successful temperature estimation has allowed creating 2D temperature maps during HIFU treatments

4:00

4pBAb4. Phase I-II study of intra-operative high intensity focused ultrasound in 25 patients with colorectal liver metastase. David Melodelima (LabTAU - INSERM U1032, 151 cours Albert Thomas, Lyon 69003, France, David.Melodelima@inserm.fr), Aurelien Dupre, David Perol, Yao Chen, Jeremy Vincenot (Ctr. Leon Berard, Lyon, France), Jean-Yves Chapelon (LabTAU - INSERM U1032, Lyon, France), and Michel Rivoire (Ctr. Leon Berard, Lyon, France)

The objective of this clinical study was to validate the effectiveness, accuracy, tolerance, and safety of a HIFU treatment developed for the treatment of liver metastases in a prospective, phase I-II trial. The transducer has a toroidal shape (diameter: 70 mm, radius of curvature: 70 mm) and was divided into 32 ring-shaped emitters operating at 3 MHz. HIFU was delivered immediately before scheduled hepatectomy. Ablations were performed on healthy tissue within the areas scheduled for resection. First, 30 ablations were carried out in 15 patients. These ablations were all generated within 40 seconds and on average measured 27.5 × 21.0 mm^2. The phase I study (n = 6) showed that use of the HIFU device was feasible and safe and did not damage neighboring tissue. The phase IIa study (n = 9) showed that the area of ablation could be precisely targeted on a previously implanted metallic mark. Ablations were achieved with a precision of 1–2 mm. The phase IIb study (n = 10) demonstrated ablation of metastases with safety margins, again prior to planned resection. Using electronic focusing the volume of ablation was adjusted to the size of the tumor from 7 cm^3 (40 s of treatment) to 50 cm^3 (6 min of treatment).

4:15

4pBAb5. A simple animal model for cerebral vasculature rupture due to exposure to intense pressure waves. Marjan Nabili, Priyanka Acharya, Yeon Ho Kim, and Matthew R. Myers (Div. of Appl. Mech., Office of Sci. and Eng. Labs.,Ctr. for Devices and Radiological Health, Food and Drug Administration, 10903 New Hampshire Ave., Bldg. 62, Rm. 2233, Silver Spring, MD 20993, marjan.nabili@fda.hhs.gov)

Understanding the threshold of microvasculature rupture in the central nervous system is critical to evaluating the safety of transcranial therapeutic ultrasound procedures, and treatment planning for blast traumatic brain injury. The goal of this study is to determine the threshold for microvasculature rupture in a simple animal model, as a function of the characteristics of the incident pressure-pulse train. An earthworm model was chosen, as a first step in a sequence of increasingly complex models, and because of its readily accessible large vessel. Following anesthetization, the earthworms were sonicated with 3.3 MHz pulse trains from a high-intensity focused ultrasound (HIFU) transducer. A variety of pulse durations, repetition rates, and amplitudes were considered. The pulse duty cycle was kept low (0.0001 to 0.001) to minimize thermal effects. In cases where rupture occurred within 10 min of exposure, the rupture time was recorded. A noticeable threshold for microvascular damage was observed at a peak negative pressure of about 20 MPa. Beyond this pressure, rupture times decreased rapidly with increasing acoustic pressure. The threshold for damage is likely due to the onset of cavitation, though the mechanisms affecting the rupture time require further study.

4:30

4pBAb6. Broadband attenuation measurements of tissue-mimicking phantoms employed for histotripsy. Michael J. Crowe (Div. of Health, Lung, and Vasculature, Dept. of Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, 3940 Cardiovascular Ctr., Cincinnati, OH 45267-0586, crowem2@xavier.edu), Jason L. Raymond (Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH), Christy K. Holland, and Kenneth B. Bader (Div. of Health, Lung, and Vasculature, Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Tissue-mimicking phantoms have previously been developed for assessing the efficacy of therapeutic ultrasound. A new modality of focused ultrasound, histotripsy, relies on the formation of shockwave-induced cavitation to ablate tissue mechanically. The objective of this study was to use a broadband ultrasound method to determine the frequency-dependent attenuation for histotripsy tissue phantoms. An agar tissue phantom was developed with evaporated milk to mimic the acoustomechanical properties of prostate tissue, a key target for histotripsy. The attenuation of this tissue phantom was measured between 2 and 20 MHz using a through-transmission technique with a single pair of PVDF transducers. The power law dependence of the acoustic attenuation spectra was linear with frequency (R^2 = 0.98). The attenuation coefficient of the tissue phantom was 0.62 ± 0.02 dB/cm/MHz, which is lower than that reported for *ex vivo* human prostate at 5 MHz. The broadband measurements techniques utilized provide a straightforward metric for the attenuation spectrum of tissue phantoms developed for shocked insonation regimes.

4:45

4pBAb7. Ultrasound causes stem cell differentiation even in the absence of specific biochemical growth factors. Nirali Shah (Mech. Eng., Swinburne Univ. of Technol., PO Box 218, Hawthorn, VIC, Melbourne, Victoria 3122, Australia), Ursula Manuelpillai (Ctr. for Reproduction and Development, Monash Inst. of Medical Res., Melbourne, VIC, Australia), Yosry Morsi, and Richard Manasseh (Mech. Eng., Swinburne Univ. of Technol., Hawthorn, VIC, Melbourne, VIC 3122, Australia, rmanasseh@swin.edu.au)

Experiments are presented on the effect of low-intensity ultrasound (LIUS) on stem cell differentiation. Previous studies had shown that LIUS can increase stem cell proliferation, and that LIUS in combination with biochemical growth factors (GFs) specific to particular cell lineages resulted in stem-cell differentiation into those lineages. However, studies had not been performed with LIUS alone. Amniotic-membrane-derived mesenchymal stem cells (MSCs) were subjected to continuous-wave insonation at 1 MHz and 0.015–0.072 MPa for periods of 10, 20, and 30 min daily for 15 days. Experiments were done both with a negative control (no insonation) and a positive control: a GF known to cause differentiation into smooth muscle cells (SMCs). The extent of differentiation was quantified by immunohistochemical fluorescence. Results showed that LIUS alone caused MSCs to differentiate into SMCs, with statistically significantly greater expression of SMC markers in the 10 and 20 min groups, with the 20 min group showing the highest differentiation. However, the 30 min group was negatively affected by insonation, with lower proliferation and differentiation. Since MSCs can differentiate into many cell lineages, the results imply that LIUS favors smooth muscle cells, with implications for *in-vivo* treatment of vascular disease and tissue engineering.

5:00

4pBAb8. Designing ultrasound fields to control the morphology of engineered microvessel networks. Eric S. Comeau (Dept. of Biomedical Eng., Univ. of Rochester, 201 Robert B. Goergen Hall, Box 270168, Rochester, NY 14627, eric.comeau@rochester.edu), Denise C. Hocking (Pharmacology and Physiol., Univ. of Rochester, Rochester, NY), and Diane Dalecki (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Spatial patterning of endothelial cells using ultrasound standing wave fields (USWF) can promote extensive microvessel network formation throughout the volume of three-dimensional collagen hydrogels. The goal of this study was to identify acoustic exposure parameters that generate specific initial spatial patterns of cells, in order to control resultant microvessel morphology. Endothelial cells were suspended in soluble collagen and exposed to a 1-MHz continuous wave USWF for 15 min during collagen gel polymerization. Samples were exposed to peak USWF pressures of 0, 0.1, 0.2, or 0.3 MPa. Samples were either imaged immediately post-exposure using high frequency ultrasound or cultured for ten days. Analysis of B-mode ultrasound images confirmed differences in initial cell band spacing and cell band density between the four exposure pressures tested. After ten days in culture, USWF-induced cell patterning resulted in three distinct microvessel morphologies. Specifically, 0.1 MPa exposure resulted in capillary-like networks, 0.2 MPa exposure resulted in non-branching vessel structures, and 0.3 MPa exposure resulted in hierarchical branching microvessel networks. Spatial characteristics of initial cell bands were then correlated with resulting microvessel network morphology. Results of these investigations allow for the capability to predictively, reproducibly, and rapidly form microvessel networks of known morphology within 3D engineered hydrogels.

5:15

4pBAb9. A nonlinear derating method for estimating high-intensity focal pressures in tissue. Seyed Ahmad Reza Dibaji (Dept. of Mech. and Mater. Eng., Univ. of Cincinnati, 2600 Clifton Ave., Cincinnati, OH 45221, dibajisa@mail.uc.edu), Yunbo Liu, Joshua E. Soneson (Office of Sci. and Eng. Labs., Ctr. for Devices and Radiological Health, U.S. Food and Drug Administration, Silver Spring, MD), Rupak K. Banerjee (Dept. of Mech. and Mater. Eng., Univ. of Cincinnati, Cincinnati, OH), and Matthew R. Myers (Office of Sci. and Eng. Labs., Ctr. for Devices and Radiological Health, U.S. Food and Drug Administration, Silver Spring, MD)

Methods for converting high-intensity focused ultrasound (HIFU) pressure measurements made in water to values appropriate for tissue have considerable value in preclinical testing and treatment planning. Such "derating" methods are straightforward in the linear-acoustics regime, but are much more difficult at higher powers. In this study, a nonlinear derating method is used to estimate focal pressure in a tissue phantom. The on-axis pressure in water was recorded using a hydrophone. Fourier transformation of the recorded pressures was performed and the resulting modal amplitudes were reduced using a combination of "source scaling" (measurements in water performed at a lower source pressure than in tissue phantom) and "endpoint scaling" (amplitudes reduced at the target location). The reduced modal amplitudes were used in the convolution term of the evolution equation to determine the pressure in tissue. The focal pressure waveform estimated by this method was compared with a direct measurement in a tissue phantom. The results show that, with the proper combination of source and endpoint scaling, focal pressure in a tissue phantom can be reproduced by derating within 15% error.

Session 4pEA

Engineering Acoustics: General Topics in Engineering Acoustics

Roger T. Richards, Chair
US Navy, 169 Payer Ln., Mystic, CT 06355

Contributed Papers

1:00

4pEA1. Aerodynamic and acoustic analysis of an industrial fan. Yi Liu (Ingersoll Rand, 800 Beaty St., Davidson, NC 28027, yi.liu@irco.com), Jeremy Bain (Bain Aero LLC, Atlanta, GA), Gang Wang (Ingersoll Rand, La Crosse, Wisconsin), Mike Lucas (Ingersoll Rand, Davidson, NC), and Percy Wang (Ingersoll Rand, Tyler, Texas)

The efforts to predict noise radiation for an industrial fan using direct computational fluid dynamics (CFD) simulation is presented in this paper. Industry has been using CFD tool to guide fan design in terms of efficiency prediction and improvement. However, the use of CFD tool for aerodynamic noise prediction is very limited in the past, partly due to the fact that research in aero-acoustics field was not practical for industry application. With the most recent technologies in CFD field, the industry application of aero-acoustics becomes much more promising. It is demonstrated here that fan tonal noise and broadband noise at low frequencies can be predicted using Overset grid system and high order finite difference schemes with acceptable fidelity.

1:15

4pEA2. An analysis of multi-year acoustic and energy performance data for bathroom and utility residential ventilation fans. Wongyu Choi, Antonio Gomez (Mech. Eng., Texas A&M Univ., Riverside Energy Efficiency Lab., 3100 State Hwy. 47, Bryan, TX 77807, wongyuchoi@tamu.edu), Michael B. Pate (Mech. Eng., Texas A&M Univ., College Station, TX), and James F. Sweeney (Texas A&M Univ., Bryan, TX)

Loudness levels have been established as a new requirement in residential ventilation standards and codes including ASHRAE and IECC. Despite the extensive application of various standards and codes, the control of loudness has not been a common target in past whole-house ventilation standards and codes. In order to evaluate the appropriate loudness of ventilation fans, especially in terms of leading standards and codes, a statistical analysis is necessary. Therefore, this paper provides statistical data for bathroom and utility ventilation fans over a nine year period from 2005 to 2013. Specifically, this paper presents an evaluation of changes in fan loudness over the 9 year test period and the relevance of loudness to leading standards including HVI and ASHRAE. The loudness levels of brushless DC-motor fans are also evaluated in comparison to the loudness of AC-motor fans. For AC and DC motor fans, relationships between loudness and efficacy was determined and then explained with regression models. Based on observations, this paper introduces a new "loudness-to-energy ratio" coefficient, L/E, which is a measure of the acoustic and energy performance of a fan. Relationships between acoustic and energy performances are established by using L/E coefficients with supporting statistics for bathroom and utility fans.

1:30

4pEA3. Evaluation and validation of the Federal Highway Administration: Traffic noise model in a sanitized environment. Monty A. Rahman (Civil Eng., Morgan State Univ., 8507 Horseshoe Rd., Ellicott City, MD 21043, rahman@mars-group.net)

The development of powerful physics based simulation models that are dependent on stochastic parameters does not always guarantee reliability. The Federal Highway Traffic Noise Model (TNM) is such a complex model that relies on Reference Energy Mean Emission Level (RMEL) from highway vehicle-type average speed and volume to predict noise propagation in far-field environment. Although the TNM accounts for static parameters such as ground type, it does not account for changing noise source or noise propagation medium. Currently, when validating TNM, the neglected dynamic parameters in the model are usually accounted for by calibrating the model to agree with measured data. However, a model is not necessarily validated when its predicted output is compared to measured data and updated to produce desired values. This paper will review current validation procedures and assess TNM's reliability to predict noise propagation from a highway line source within a sanitized environment. Sanitized environment is an open field within which noise propagation is independent of dynamic variables. By validating TNM within a sanitized environment TNM's residual prediction error due only to variation in the traffic parameters (vehicle type, speed, and flow rate) could be identified and the reliability of REML used in TNM could be confirmed.

1:45

4pEA4. An improved two-microphone transfer function method for measuring oblique absorption coefficient in the free field using numerical modeling. Hubert S. Hall (Signatures, Naval Surface Warfare Ctr. Carderock Div., 620 Michigan Ave. NE, Washington, DC 20064, 61hall@cardinalmail.cua.edu), Joseph F. Vignola, John A. Judge, Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

The use of the two-microphone free field transfer function technique has remained unchanged since its development. In practice, finite sample sizes limit measurement fidelity due to edge effects. The sound field contribution from edge diffraction has generally restricted the technique from use at low frequencies (<300 Hz) and required test panels greater than several square meters in area. This effort intends to characterize the diffraction contribution to the sound field for relatively small panels. Using numerical modeling, samples of like acoustic properties were excited by a point source at normal incidence to quantify the diffraction term. Following validation via comparison with impedance tube data, the diffraction term was incorporated in an updated derivation of the complex reflection coefficient equation and validated experimentally for a known reference material at both normal and oblique incidence. The goal of this work is to validate measurements of oblique absorption coefficient in an anechoic chamber for panels smaller than one square meter at frequencies down to 100 Hz.

2:00

4pEA5. Initial laboratory experiments to validate a phase and gradient estimator method for the calculation of acoustic intensity. Darren K. Torrie, Eric B. Whiting, Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N223 ESC, Provo, UT 84602, darren@torriefamily.org)

A recently developed phase and gradient estimator (PAGE) method for calculating acoustic intensity from multiple pressure measurements [Thomas *et al*., J. Acoust. Soc. Am. **134**, 4058 (2013)] has been tested via anechoic laboratory measurements of the radiation from multiple loudspeakers. The measurements were used to examine the effects of probe geometry, size, frequency range, and source characteristics on the active intensity calculated from both the PAGE and the traditional cross-spectral approaches. Preliminary results are shown for multiple probe and source configurations, and confirm that the PAGE method results in a broader valid frequency range for a given probe geometry.

2:15

4pEA6. Reverberation mapping using instantaneous intensity estimates from a tetrahedron microphone array. Thomas Burns (Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344, tburns@starkey.com)

For hearing aids, the directivity index is a benchmark defined under two acoustical conditions that a hearing-aid user won't encounter, namely, the ratio of sound power signal arriving from the on-axis target in an anechoic condition to the isotropic spherical noise. It would be useful to benchmark the speech signal to noise that is encountered in a typical environment. The purpose of this study is to map the instantaneous acoustic intensity in a room using a head and torso voice simulator as the source and a regular tetrahedron microphone array in the field. Four impulse responses from a small tetrahedron were measured in a reverberant conference room of gypsum walls, carpet, and absorptive ceiling tile. Welch's method was used to compute one-third octave estimates of the auto- and cross-spectra from the impulse responses, and these spectra were used to estimate the steady-state, 3D instantaneous acoustic intensity vector via the time-averaged active intensity and the maximum amplitude of the reactive intensity. A histogram of the instantaneous intensity vector revealed an angular estimate for the arriving sound power that was, in general, cylindrically rather than spherically "diffuse."

2:30

4pEA7. Acoustic characterization of micro-perforate porous plates. Martha C. Brown (AeroAcoust. Branch, NASA Langley Res. Ctr., 1D East Reid St., Rm. 105C, M.S. 164D, Hampton, VA 23681, martha.c.brown@nasa.gov), Michael G. Jones, and Brian M. Howerton (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

The research objective of this paper is to develop an acoustic impedance model for micro-perforate plates. Passive acoustic liners consisting of perforate plates-over-honeycomb structures are a key contributor in the reduction of fan noise propagated through the inlet and aft-fan duct of aircraft engine nacelles. These perforated plates are physically characterized by plate thickness, t, orifice diameter, d, and porosity, σ, and the resultant liners are typically classified as conventional for $t/d \approx 1$ or micro-perforate for $t/d \gg 1$. Micro-perforate plates are becoming more popular because of their acoustic linearity, i.e., insensitivity to sound level. The goal of the current research is to develop models to better understand the acoustic behavior of liners constructed with micro-perforate facesheets. The first step is to model the flow resistivity of micro-perforate plates, computed as the ratio of the static pressure drop across the plate to the mean flow through the orifices. This result is then used to model the acoustic impedance, which is defined as the ratio of the acoustic pressure to the normal component of acoustic particle velocity at the liner surface. These tests are conducted in the NASA Langley Raylometer and Normal Incidence Tube with samples spanning a range of $4\% \leq \sigma \leq 20\%$ and $5.0 \leq t/d \leq 7.14$.

2:45

4pEA8. Experimental tests of protective covers to obtain optimal transmission loss in porous materials. Jin Liu (UTC Carrier Technol., Methods & Components Eng., 107 Woodmancy Ln., Fayetteville, NY 13066, lewjin@yahoo.com) and John Wang (UTC Carrier Technol., Methods & Components Eng., Syracuse, NY)

Porous resistive materials are widely used to increase the wide-band transmission loss and reduce shell vibrations and noise propagation. For industrial applications, porous materials need to be covered to protect them from environmental contaminants. The goal of our experimental studies is to determine suitable covers which are acoustically transparent and offer this needed protection. Expansion volumes were designed for transmission loss tests with two accurate modeled limits related to two rigid boundary conditions. Seven different covers were tested in this study tests. Best results were achieved using thin porous steel woven cloth. Tyvek, Mylar, thick steel woven cloth, and the Nomex were not as effective. The porous materials and covering materials were tested and then retested after being soaked in oil to explore transmission loss that might occur in actual industrial conditions. These independent measurements of the cover materials were used to determine the material property in the FEA models.

3:00

4pEA9. Unconditionally stable time-domain elastic wave simulations by the alternative direction implicit method. Yu Shao, Myoung An, and Shumin Wang (Auburn Univ., 200 Broun Hall, Auburn, AL 36849, wangs@auburn.edu)

The stability of the conventional staggered-grid finite-difference time-domain (FDTD) method for elastic wave simulations is limited by the Courant condition and material heterogeneity. Its computational efficiency is significantly hampered when the mesh size is much smaller than a wavelength (for geometric modeling accuracy) and/or with a high impedance contrast. An unconditionally stable alternating direction implicit (ADI) method is proposed to overcome the conditional stability. It is based on additively splitting the Crank Nicholson (CN) operator into two sub-operators and subsequently solving the CN scheme in two sub-steps. In each sub-step, a tri-diagonal matrix is formed based on one of the sub-operators and the associated field variables are solved implicitly with O(N) computational complexity. The rest of the field variables are solved explicitly. Due to its semi-implicit nature, it can be proved that the ADI method is unconditionally stable regardless of the time step size. The numerical dispersion properties of the ADI method are also analyzed theoretically. Several numerical examples of acoustic wave scattering from elastic targets are further provided to demonstrate its accuracy and efficiency.

3:15–3:30 Break

3:30

4pEA10. Movable optical lens array using acoustic radiation force. Daisuke Koyama, Yuta Kashihara (Faculty of Sci. and Eng., Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe, Kyoto 610-0321, Japan, dkoyama@mail.doshisha.ac.jp), Megumi Hatanaka (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan), Kentaro Nakamura (Precision and Intelligence Lab., Tokyo Inst. of Technol., Yokohama, Japan), and Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Japan)

A movable optical lens array that utilizes acoustic radiation force was investigated. The lens array consists of a rectangular glass plate, two piezoelectric bimorph transducers, and a transparent viscoelastic gel film. By electrically exciting the transducers, the flexural vibration mode was generated along the glass plate. The acoustic radiation force acts to the surface of the gel so that the surface profile can be deformed and the lens array can be fabricated. A cylindrical lens array with the lens pitch of 4.6 mm was fabricated at 113 kHz. The lens positions correspond with the loop positions of the flexural standing wave, and the lens pitch corresponds with half the

wavelength of the flexural vibration of the glass plate. The focal points of the lens could be changed by the input voltage, and the lens can act as a variable-focus lens. The lens positions could be moved in the length direction by two-phase drive. The moving distance of the lens was 4.6 mm when the driving phase difference between the two transducers changed from 0 to 360°. The translation of the lens array was applied as an optical scanner.

3:45

4pEA11. Thermoacoustic energy harvesting. Andrew W. Avent and Christopher R. Bowen (Mech. Eng., Univ. of Bath, Claverton Down, Bath, Sommerset BA2 7AY, United Kingdom, a.avent@bath.ac.uk)

Thermoacoustics have a key role to play in energy harvesting systems, exploiting a temperature gradient to produce powerful acoustic pressure waves. As the name suggests, thermoacoustics is a blend of two distinct disciplines: thermodynamics and acoustics. The field encompasses the complex thermo-fluid processes associated with the compression and rarefaction of a working gas as an acoustic wave propagates through closely stacked plates in the regenerator of a thermoacoustic device; and the acoustic network that controls the phasing and properties of that wave. Key performance parameters and appropriate figures of merit for thermoacoustic devices are presented with particular emphasis upon the critical temperature gradient required to initiate the acoustic wave and the thermal properties of the key component, namely, the "stack" or "regenerator". Mechanisms for coupling a thermoacoustic prime mover with electromagnetic harvesters and piezoelectric transducer materials are also presented, which offer the potential to enhance the energy density attained beyond that possible with linear alternators. Numerical modeling strategies are presented, which enable parametric sweeps of the geometric and thermal properties, which influence the efficiency, and performance of the key components of such devices. Potential coupling and non-linear effects are examined.

4:00

4pEA12. A unified equivalent circuit of electromechanical transducers. Li-Feng Ge (School of Elec. Eng. and Automation, Anhui Univ., 111 Jiulong Rd., Hefei 230601, China, lf_ge@hotmail.com)

It has been interesting to find a unified equivalent circuit of electromechanical transducers for a long time. Hunt provided a method using a space operator to represent all transducer types with a single form of equivalent circuit [Hunt, Electroacoustics, 1954]. But, as indicated by Beranek, the space operator k does not commute with the time operator j, so that one must define $jk=-kj$ [Beranek, JASA **77**(4), 1309–1313, 1985]. During the last decades, the research on micromachined electrostatic or capacitive ultrasonic transducers stimulates further the interest in the subject [Ge, Sc. China, A **29**(11), 1308–1315, 1999]. In this work, the space operator is not used. A transformation factor is defined as a ratio of blocked electrical-to-mechanical transfer impedance to blocked electrical impedance, thus, the factor is a real number for reciprocal transducers and imaginary for antireciprocal ones, which indicates the essential difference of the two types of transducers. Then, the corresponding driving-point electrical impedances are derived, and the expressions for the two are identical, demonstrating their common properties also. Thus, a unified impedance-type equivalent circuit for electromechanical transducers can be further obtained. [Work supported by NSFC (Grant Nos. 60374044 and 60774053).]

4:15

4pEA13. Fabrication of micro-lens array using surface acoustic wave. Satoki Taniguchi (Faculty of Life and Medical Sci., Doshisha Univ., Tataramiyakodani 1-3, Kyotanabe, Kyoto 610-0321, Japan, dmo1038@mail4.doshisha.ac.jp), Shinji Takayanagi, Daisuke Koyama (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan), Kentaro Nakamura (Precision and Intelligence Lab., Tokyo Inst. of Technol., Yokohama, Japan), and Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Japan)

A technique to form an optical micro-lens array using surface acoustic wave (SAW) was investigated. The lens has no mechanical moving parts, such as gearing systems, and is composed of a simple structure. A visco-elastic transparent silicone gel film with the thickness of 20 μm was formed on a 128°-rotated Y-cut X-propagation $LiNbO_3$ (LN) substrate between two interdigitated transducers (IDTs). The IDT electrodes consisted of 30 finger pairs with the aperture length of 9.1 mm and the periodic pitch of 200 μm were fabricated on the LN substrate. The IDTs was excited with a continuous sine wave with the amplitude of 0 to 15 V_{pp} at the resonance frequency (20 MHz) to generate the SAW. The leaky SAW propagated into the gel, and the acoustic radiation force acted to the surface of the gel film so that the surface profile of the gel could change and a micro-lens array could be fabricated on the gel. The lens height could be controlled by varying the voltage applied to the IDTs.

4:30

4pEA14. *PT*-symmetric acoustics. Xuefeng Zhu, Hamidreza Ramezani, Chengzhi Shi, Jie Zhu, and Xiang Zhang (Dept. of Mech. Eng., Univ. of California, Berkeley, 3112 Etcheverry Hall, Berkeley, CA 94720, chengzhi.shi@berkeley.edu)

The concept of acoustic parity-time (*PT*) symmetry is introduced and used for the study of extraordinary scattering behavior in acoustic *PT*-symmetric media consist of loss and gain units. The analytical study of acoustic *PT*-symmetric media shows that these media can be designed to achieve unidirectional transparency at specific frequencies named exceptional points (EPs). This unidirectional transparency occurs at the EPs is due to the asymmetrical arrangement of the periodic loss and gain units that results in different Bragg scatterings on the two sides of the *PT*-symmetric media. A close look at the phases of the reflections on both sides reveals a sudden jump of the reflection phase on one side at the EPs. This step-function like behavior causes an infinite delay time of the reflected wave on that side, and hence the media become reflectionless in that direction. Combining the concept of acoustic *PT*-symmetry with transformation acoustics, we design a two-dimensional acoustic cloak that is invisible in a prescribed direction. This kind of directional cloak is important especially for military use since a target object is hidden from the enemy in front can still be identified by friendly at the back. Other useful applications such as directional acoustic imaging, noise cancellation, architectural acoustics, acoustic amplification, etc., can also be developed.

4:45

4pEA15. Smoothed particle hydrodynamics approach for modeling sound of a rigid body falling on water. YongOu Zhang, Tao Zhang, TianYun Li, and Peng Wang (Dept. of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., East 2nd Bldg., Wuhan, Hubei 430074, China, zhangyo1989@gmail.com)

Computational acoustic methods based on Eulerian description are widely used in industrial applications. However, some special acoustic problems, such as transient acoustic problems with moving or deformable boundaries, object separation, or for multiphase systems, are still cannot ideally solved with these Eulerian methods. The present work aims at using a Lagrangian meshfree method, the smoothed particle hydrodynamics (SPH), to simulate a time-domain acoustic problems with moving boundaries which is the sound of a rigid body falling on water. First, Lagrangian acoustic wave equations considering the sound source based on the hydrodynamic/acoustic splitting method are given and represented in the SPH form. Then, two-dimensional simulation of a rigid object falling on the free surface of water is computed by the SPH method. Noise sources are obtained from the flow field information of each fluid particle. Finally, acoustic experiments with measuring the sound of the rigid body falling on water are used to test and validate the SPH results. The accuracy and efficiency of the SPH acoustic computation are evaluated, and a comparison of cases with different impact velocities is also discussed.

5:00

4pEA16. Identification of fractures in carbonates using sonic imaging logs Example from the central of East European plain. Xiaohua Che, Wenxiao Qiao, Peng Liu, Xiaodong Ju, and Junqiang Lu (State Key Lab. of Petroleum Resources and Prospecting, College of Geophys. and Information Eng., China Univ. of Petroleum-Beijing, 18 Fuxue Rd., Changping, Bejing 102249, China, imchexh@gmail.com)

Compared with sandstone, the structure of carbonate rocks is much more complex for its heterogeneity and anisotropy. The Republic of

Tatarstan of Russia lies in the central of East European Plain, where oil-gas resource is very rich, and formations are mainly carbonates. In order to detect the underground geology of the region, array acoustic logging tool, micro-electrical scanning tool, and conventional logging tools were applied. We processed and analyzed the array acoustic logging data. The results demonstrated that monopole shear wave attenuation, Stoneley wave attenuation and monopole variable-density waveforms were all sensitive to the boundary where wave impedance contrast was high. The boundary may be a fracture, or a bed interface between shale and limestone in the region. In the micro-electrical scanning images, both the low-angle fracture and the bed interface showed one sine curve. The sine curve of the low-angle fracture was relative rough and the curve width changed with different azimuth, compared with that of the bed interface. The direction of fracture and stress can be obtained by processing cross-dipole waveforms of array acoustic logging data. Micro-electrical scanning tool provided three caliper curves of different azimuth that can determine borehole expanding direction, further getting the stress direction.

THURSDAY AFTERNOON, 21 MAY 2015 BRIGADE, 1:30 P.M. TO 4:30 P.M.

Session 4pMU

Musical Acoustics: General Topics in Musical Acoustics II

Randy Worland, Cochair
Physics, University of Puget Sound, 1500 N. Warner, Tacoma, WA 98416

Jack Dostal, Cochair
Physics, Wake Forest University, P.O. Box 7507, Winston-Salem, NC 27109

Whitney L. Coyle, Cochair
The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Chair's Introduction—1:30

Contributed Papers

1:35

4pMU1. Measurement and analysis of musical vibrato parameters. Mingfeng Zhang, Mark Bocko (Dept. ECE, Univ. of Rochester, Rochester, NY 14627, mzhang43@hse.rochester.edu), and James W. Beauchamp (ECE, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Vibrato is a fundamental feature of many musical sounds, with vocal, stringed instrument, and many wind instrumental performances employing vibrato as a principal device for conveying musical expression. In this paper, we describe a signal processing toolbox for extracting and analyzing vibrato-related parameters from audio recordings. In our method, sonic partials of a musical sound are separated first, followed by tracking of the instantaneous amplitude and frequency of each partial. A number of parameters that characterize vibrato can then be extracted from the tracked partials; these include the common frequency of the amplitude modulation (AM) and the frequency modulation (FM), the modulation depths, the relative phase between the AM and FM components of each partial, and the harmonic contents of the modulation components. This then enables a detailed comparison of the vibrato of various musical sounds, for example the relative amounts of AM and FM, the rate of vibrato, the correlation and phase between AM and FM vibrato components, and many other features. This framework provides a useful tool for applications such as music performance pedagogy and musicological studies.

1:50

4pMU2. Real-time visualization of musical vibrato for music pedagogy. Minhao Zhang, Mingfeng Zhang, Sarah Smith (ECE, Univ. of Rochester, Rochester, NY 14627, mzhang46@ur.rochester.edu), and Mark Bocko (ECE, Univ. of Rochester, Rochester, NY)

Vibrato is a fundamental expressive attribute in music, especially in singing, in stringed instrument performance, and in the performance techniques of many wind instruments. Performers typically invest a great deal of time and practice to gain adequate control of vibrato in performance To assist and accelerate this learning process we are developing computer based vibrato visualization tools. In this paper, we present a low-latency, near real-time system that enables performers to visualize their vibrato. The system employs a video-game-like visualization scheme to display the instantaneous AM/FM trajectories of a musical sound, either by itself or in comparison to pre-recorded target sounds, to enable the user to attempt to match the vibrato trajectory of an existing performance. In addition to demonstrating the system, feedback from music students and studies of the effectiveness of the system will be presented.

2:05

4pMU3. Listener preferences for vibrato rate and extent in synthesized vocal samples. John P. Nix (Music, Univ. of Texas San Antonio, One UTSA Circle, San Antonio, TX 78249, john.nix@utsa.edu)

An experiment was devised to test the preferences of singers, vocal teachers and choral directors for vibrato rate and extent on the /a/ and /i/ vowels and with two different "gender" singers by using the voice synthesis program Madde, which is capable of varying vibrato rate, extent,

fundamental frequency, and formant frequencies. The study also sought to determine if the vowel and "gender" of the singer (as determined by the formant frequencies used for the simulation) have an impact on listener preferences. Subjects answered a series of questions about their musical training and occupation, then listened to four samples each, two of the "male" voice performing the vowels at 220 Hz, and two of the "female" voice performing the same vowels at 440 Hz. The order in which the samples were presented and the initial vibrato rate and extent conditions of the simulations were varied to try to negate any order effects on the data. The investigator adjusted the vibrato rate and vibrato extent until the listener felt his or her preference was being best represented. Listeners displayed a wide variety of preferences, with those most involved in musical theater and early music performance preferring a narrower vibrato extent.

2:20

4pMU4. Perception of non-vibrato sung tones. Randi Wooding and John P. Nix (Music, Univ. of Texas San Antonio, One UTSA Circle, San Antonio, TX 78249, john.nix@utsa.edu)

Singers are often asked to sing with a non-vibrato production. However, the term non-vibrato is problematic, as all human singing involves fundamental frequency variation. Whether a singer achieves a quality of tone that is perceived as non-vibrato may depend upon the experience of the listener. The specific aim of this study was to determine at what point a tone is perceived as non-vibrato by a population (N = 131) of singers, voice teachers, choir directors, and speech pathologists. Utilizing voice samples that exhibit a variety of vibrato rates and extents, the investigators sought to determine: (1) if there is a threshold for the perception of non-vibrato tone with regards to vibrato extent; (2) if given similar vibrato extent, does vibrato rate effect perceptual judgments of non-vibrato tone; (3) if there are differences in the perceptual threshold of non-vibrato tone across different professions of listeners. Participants responded to an online survey featuring randomized samples of sopranos singing [ɑ] with a variety of vibrato rates and extents. Some samples were repeated to test response reliability. Results indicate a perceptual threshold exists with regards to vibrato extent, and that vibrato rate can override the effect of a narrow or wide extent to some degree.

2:35

4pMU5. Estimation of a violin source/filter model. James W. Beauchamp (Dept. of Elec. and Comput. Eng. and School of Music, Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr., Urbana, IL 61801-6824, jwbeauch@illinois.edu) and Ashwin K. Vijayakumar (Dept. of Electronics and Commun. Eng., National Inst. of Technol., Karnataka, Surathkal, Karnataka, India)

The source/filter model is often mentioned in the acoustics and signal processing literature (e.g., Gold and Morgan, *Speech and Audio Signal Processing*, Wiley) but has seldom been implemented for musical instrument sounds. An exception was by Mathews and Kohut, ("Electronic simulation of violin resonances," J. Acoust. Soc. Am. **53**(6) (1973)). In this study, we attempt to separate source harmonic amplitudes from a violin filter characteristic using harmonic tracking analysis of glide tones performed in an anechoic chamber. Assuming that the violin bridge/body constitutes a linear system, if string force harmonics are constant over pitch, they should sweep out frequency responses that only differ by constant amplitude ratios over their overlap regions. Departures from constant ratio behavior can be interpreted in terms of source spectra that change with fundamental frequency. Interpretation and detailed results of this analysis are given.

2:50

4pMU6. The influence of sung vowels on pitch perception. Johanna Devaney and Derek Richardson (School of Music, The Ohio State Univ., 1899 N. College Rd., Columbus, OH 43210, devaney.12@osu.edu)

Phonologists have demonstrated that in speech there is an intrinsic pitch of vowels, i.e., that different pitch heights are perceived for different spoken vowels with the same mean fundamental frequency. Only limited work has been done on the impact of sung vowels on perceived pitch and previous studies, using forced choice paradigms, have found conflicting results comparing Front/Close to Back/Open vowels: Ternstrom, Sundberg, and Colldén (1988), using manipulated sung tones, found that Front/Close vowels were perceived as higher than other vowels while Fowler and Brown (1997), using synthesized tones, found that Front/Close was perceived lower than Back/Open. The current study uses a method-of-adjustment paradigm to examine pitch perception of all four extremal vowels: Front/Close (/i/), Front/Open (/æ/), Back/Close (/u/), and Back/Open (/ɑ/), using completely controlled, but realistic synthesis. It also assesses whether different models of pitch perception are necessary for the various vowel types. In the experiment, the subjects are asked to match a reference tone that has no formants against a stimulus tone with first and second formants corresponding to a particular vowel. Both the reference and stimulus tones are identical in their synthesis except for the formants and are 300 ms in length.

3:05–3:15 Break

3:15

4pMU7. Effect of acoustic feedback on the singing voice. Pasquale Bottalico (Dept. of Communicative Sci. and Disord., Michigan State Univ., Via Sant'Anselmo 32, Torino 10125, Italy, pasqualebottalico@yahoo.it), Eric J. Hunter, and Simone Graetzer (Dept. of Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Voice control is of great importance in a singer's training, and in particular, control of pitch (fundamental frequency) and volume (sound pressure level). We hypothesize that a singer's vocal comfort and control will be increased as the acoustic feedback of the room is increased. In this study, 20 singers (10 amateur and 10 professional singers) performed vocal tasks in different acoustic conditions. Vocal tasks comprised scales and arpeggios with different dynamics at different speeds, and an extract from the American National Anthem, which was accompanied by a musical track emitted at different power levels. After each condition, the subject responded to questions regarding perception of vocal comfort, control, and fatigue, and their own voice feedback. Room acoustics were manipulated: in some conditions, two reflective panels were placed at different distances from the subject. The results indicate that when panels were present, singers tended to perceive the room as being more pleasant to sing in. Intonation (pitch control) and the Lombard Effect (a tendency to increase in voice level as background noise level increases) for singers are related to variation in the room acoustics.

3:30

4pMU8. Differentiated electroglottograph and audio signal measurements of vocal fold closed quotient during a register change: Single note data. Richard J. Morris (Commun. Sci. and Disord., Florida State Univ., 201 West Bloxham Rd., 612 Warren Bldg., Tallahassee, FL 32306-1200, richard.morris@cci.fsu.edu), David Okerlund (College of Music, Florida State Univ., Tallahassee, FL), and Shonda Bernadin (College of Eng., FAMU/Florida State Univ., Tallahassee, FL)

Recently, the use of the differentiated electroglottograph (dEGG) signal combined with a differentiated audio (dAudio) signal was reported as a means for more reliable measurement of the closing quotient (CQ) from the EGG signal in combination with a time synchronized audio signal. The purpose of this study is to combine CQ data with resonance data from the vocal tract during a register shift. Files recorded from group of 15 trained females provided the data. The singers were directed to shift between chest and mixed registers while sustaining a note that could be sung comfortably in both registers. An EGG and an audio signal were recorded. These signals were time aligned, and the EGG, dEGG, and dAudio were displayed. These three signals were used to determine the CQ and peak resonant amplitude before and after the register shift. Preliminary results indicate that the singers maintained a similar CQ measurement across the register shift. In addition, the preliminary data indicate that the singers changed the dominant harmonic in the sung note when they changed registers. That is, they maintained the same fundamental frequency but altered the resonance of the tone in the vocal tract.

3:45

4pMU9. Comparing time varying directivity of musical instruments across different musical motifs. Madeline A. Davidson, Kristin Hanna, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, madeline.davidson@huskers.unl.edu)

The directionality of a musical instrument can vary in time quite significantly as the instrument is used to play a musical piece, but this behavior has not been well-quantified or compared across instruments and musical motifs in previous research. Prior work conducted at the University of Nebraska has proposed quantifying this behavior by seeing how directional characteristics vary in time using overlapping one-second windows. The analyses were conducted on twenty-second long sections from 13-channel anechoic recordings of three solo instruments (flute, violin, and trombone). Maximum directivity index plots across time and space were produced to quantify and examine the time-varying directional characteristics. In this presentation, these previous results are compared against more recent results obtained from five-channel anechoic recordings of all instrument parts of both a Mozart and a Brahms symphony. The effects of reducing the time window of analysis from one second down to half-second and quarter-second time windows are also examined. Special attention is paid to how the proposed quantifiers for time-varying directivity are impacted by aspects such as tempo and dynamics in these musical compositions.

4:00

4pMU10. Effects of pitch, intensity, and timbre on frequency masking. Song Hui Chon and David Huron (Ohio State Univ. School of Music, 1866 College Rd., Columbus, OH 43210, chon.21@osu.edu)

It has been widely known that the four main factors of music—pitch, timbre, duration, and loudness—are not independent. One comes to wonder how these four factors will affect frequency masking, which has traditionally been explained with the cochlear responses to pure tones or narrowband noises. As a first step toward understanding these interactions, the mutual masking effects of two concurrent non-unison tones were investigated. A MATLAB simulation was implemented based on the psychoacoustic model in Bosi & Goldberg (2003). As the degree of masking would obviously depend on factors such as pitch, timbre, and intensity, we considered 81 pitches (B0 to G7) in three timbre categories (pure tone, uniform, and average timbre). For each timbre category, 15 levels of amplitudes were considered to examine the effect of intensity. A preliminary analysis shows that the chance of the higher pitched tone masking the lower pitched tone might change depending on the pitch, timbre, and loudness. The findings will be discussed with the implications on the interpretation of high voice superiority on the preference of melodies on the highest part in music.

4:15

4pMU11. Demonstration of nonlinear mode coupling in woodwind-like air columns: Recollections from the laboratory of A. H. Benade. Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ., N-249 Millennium Sci. Complex, University Park, PA 16803, sct12@psu.edu) and Peter L. Hoekje (Phys. and Astronomy, Baldwin Wallace Univ., Berea, OH)

Self sustained oscillations in woodwinds are generated by the player's blowing pressure, through the interaction of the standing wave in the air column, the dynamic motion of the reed and the air flow through the reed. In properly designed instruments, the oscillation begins at a threshold blowing pressure near the frequency of an air column resonance. With increased blowing pressure, the amplitude grows and the flow nonlinearity generates harmonics of the playing frequency. The nonlinearity couples the steady state components of the tone. The coupling dictates behaviors that cannot be explained by a linear model. When A. H. Benade came to understand this in the early 1960s, he developed a "tacet horn" whose unplayable behavior could only be explained by the nonlinear theory [Benade and Gans, Ann. N.Y. Acad. Sci. **155**, 247–263 (1968)]. Throughout his life, Benade used other pathological air columns for research and pedagogy. This paper will demonstrate two of Benade's concepts with air columns that are simple to build. One is a clarinet-like tube that can play softly near threshold, but cannot be played louder. The other is a conical tube that plays only at high amplitude.

THURSDAY AFTERNOON, 21 MAY 2015 — KINGS 5, 2:00 P.M. TO 5:00 P.M.

Session 4pNS

Noise, ASA Committee on Standards, Psychological and Physiological Acoustics, Physical Acoustics, and Structural Acoustics and Vibration: Wind Turbine Noise II

Robert D. Hellweg, Cochair
Hellweg Acoustics, Wellesley, MA 02482

Kenneth Kaliski, Cochair
RSG Inc, 55 Railroad Row, White River Junction, VT 05001

Paul D. Schomer, Cochair
Schomer and Associates Inc., 2117 Robert Drive, Champaign, IL 61821

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

This session builds on the session 4aNS at which research and technical background useful toward the development of the wind turbine acoustic emissions agenda will be presented. Session 4pNS will consist of panelists along with audience participation with the purpose of developing the start of a draft research agenda. Invitees, to the best of our abilities, will include all interested parties, including relevant agencies from the United States and Canada, medical practitioners, other research professionals, and acoustical consultants with wind turbine acoustic emissions experience.

Session 4pPA

Physical Acoustics: Infrasound II

Roger M. Waxler, Chair
NCPA, University of Mississippi, 1 Coliseum Dr., University, MS 38677

Contributed Papers

2:00

4pPA1. Propagation model based explosive yield determination from stratospheric infrasound arrivals: Humming Roadrunner data analysis. Roger M. Waxler (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu), David Green (AWE Blacknest, Reading, United Kingdom), and Jean-Marie Lalande (NCPA, Univ. of MS, University, MS)

Attempts to estimate explosive yields from stratospheric infrasonic returns have often relied on empirical relationships between explosive charge weight, source-to-receiver range, and a simple parameterization of the atmospheric state. We propose to approach the yield estimation problem by combining knowledge of acoustic generation by explosions and numerical acoustic propagation modeling using state-of-the-art atmospheric specifications (G2S). The goal is to provide a more accurate and robust yield estimation tool that can incorporate a variety of meteorological scenarios. We introduce an approach based on modeling the transmission loss between source and receiver using an incoherent modal sum. The total observed signal power is related to the near-field acoustic power generated by the source event. The technique is applied to stratospheric returns recorded from the Humming Roadrunner Ground Truth experiments performed in the August 2012 in the American West.

2:15

4pPA2. Theoretical model and experimental validation of a gas-combustion infrasound source. Chad M. Smith and Thomas B. Gabrielson (Graduate Program in Acoust., Appl. Res. Lab., The Penn State Univ., State College, PA 16804, cms561@psu.edu)

The design of controlled infrasound sources is challenging due to the large volumes of air that must be moved with a source of practical size. One possible approach is to produce a periodic volume expansion/contraction by periodic heating of the air mass using a combustible-gas burner. For equal storage pressure, the available energy density from combustion of a gas is substantially higher than the available energy density from expansion of a compressed gas. While no source that is much smaller than a wavelength will be anything but inefficient, this large energy density may partially offset this basic acoustical inefficiency making gas-combustion infrasound source technology practical for use in the field. The goal of this work is to generate usable levels of sound in the 0.1 to 20 Hz frequency band for infrasound-measurement-system calibrations and propagation-path characterization. This paper presents a theoretical model and experimental demonstration of this infrasound generation method. Both theory and demonstration suggest that usable infrasound levels can be generated with a relatively compact source.

2:30

4pPA3. The validation of SSWs forecasts in atmospheric global circulation models. Jelle D. Assink (DAM/DIF/DASE, CEA, Bruyeres-le-Chatel, Arpajon F-91297, France, jelle.assink@gmail.com), Pieter Smets, Läslo Evers (KNMI, De Bilt, Netherlands), and Alexis Le Pichon (DAM/DIF/DASE, CEA, Arpajon, France)

While the influence of the troposphere on the stratosphere is well known, recent observational and modeling studies have demonstrated that the stratosphere has an impact on the troposphere as well. The dynamical coupling between stratosphere and troposphere is particularly strong during sudden stratospheric warming (SSW) events. The correct forecasting of the onset and duration of SSW events is therefore important and is a current challenge for weather forecasting centers. As there is a lack of observations in the upper stratosphere with good temporal and spatial coverage, additional techniques may be helpful to constrain SSWs. This is illustrated using volcanic infrasound measurements. The observations are compared with nowcast and forecast models up to 10 days. While a general agreement is found during the summer period, larger discrepancies are found during the equinox and major SSW of January 2013.

2:45

4pPA4. Status of digital infrasound sensors developed by the NCPA. Carrick L. Talmadge (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38655, clt@olemiss.edu)

In collaboration with Hyperion Technology Group, Inc., the National Center for Physical Acoustics (NCPA) has developed a digital infrasound sensor that can be configured for broadband outdoor measurements (flat within 3-dB from 0.03–150 Hz) and a nominal maximum transducible pressure of 200-Pa peak-to-peak, ultra-broadband measurements for calibration systems (flat within 3-dB from 0.0003-Hz to 150 Hz) and very high level sounds (up to 110 kPa peak-to-peak). This sensor has a GPS-locked digitizer that store over four months of continuously sampled data digitized at 1000 samples per second. The measurement performance of this system, including noise floor, reproducibility of measurements between sensors, linearity, mechanical robustness, etc., will be summarized.

4p THU. PM

3:00

4pPA5. The interaction between infrasonic waves and gravity waves perturbations: Application to explosions at the Utah test and training range. Jean-Marie Lalande and Roger Waxler (National Ctr. for Physical Acoust., Univ. of MS, 1 Coliseum Dr., University, MS 38677, jlalande@olemiss.edu)

Infrasonic waves propagate at long range through atmospheric ducts resulting from the stratification of atmospheric properties. These ducts are characterized by their spatio-temporal variability. Hence, infrasonic waves integrate information upon the atmosphere along their propagation paths. In order to study infrasonic wave propagation, we resort to atmospheric specification combining numerical weather prediction and climatological models. However, due to the lack of observations these models fail to describe small scale variability such as perturbations associated to the presence of internal

gravity waves. These waves play an important role in the atmospheric dynamic by transferring momentum to the mean flow at critical levels and at wave-breaking altitudes. In this study, we intend to describe the interaction of infrasonic waves with internal gravity waves in order to understand the long-tail behavior observed in infrasound broadband signals. We developed a model for the propagation of internal waves used to generate realistic perturbations of the background atmospheric states. By using a linear full-wave model of infrasound propagation, our goal is to ultimately relate infrasound characteristics to internal waves properties. We apply those numerical models to infrasound observations recorded during the 2010 deployment of the Utah Test and Training Range (UTTR) experiment.

3:15

4pPA6. On atmospheric updating, attenuation, and thermospheric phases in infrasonic wave-trains. Roger M. Waxler (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu), Jelle Assink (CEA, Paris, France), and Joel Lonzaga (NCPA, Univ. of MS, University, MS)

Because of the dramatic increase in temperature in the thermosphere, thermospheric phases are ubiquitous, despite the fact that they are less often detected than are stratospheric phases. In the thermosphere, the density of the atmosphere is very low so that, as a propagation medium, the thermosphere is very non-linear and very attenuating. Atmospheric specifications of the lower thermosphere are not well constrained by data. This can lead to significant errors in modeling propagation along thermospheric paths. Methods for generating corrections to specifications of the thermosphere from infrasonic data are described. Further, using a non-linear ray theory model we find that non-linear effects in the thermosphere, in particular, period lengthening, mitigate against attenuation. Finally, we show that it is possible to use thermospheric returns to simultaneously update the atmospheric specifications while determining source yield.

3:30

4pPA7. Infrasonic propagation modeling of orbital launch vehicles. John M. Noble, W. C. K. Alberts, and Stephen M. Tenney (Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, john.m.noble.civ@mail.mil)

Over the last two years, NASA's Wallops Flight Facility has been launching medium lift rockets for experimental and space station resupply missions. These launches have been a great opportunity to use the rocket-generated infrasound as a repeatable source to study the long range propagation over different seasons. For some of the launches during this period, two, 20-m arrays were deployed along different directions from the launch site. To increase the number of available sensors for comparison to three-dimensional propagation model results, data from the US Array was incorporated into the study. The US Array data significantly increased the range and azimuths used to compare the propagation modeling and measurement results. Calculated and measured transmission losses will be discussed.

THURSDAY AFTERNOON, 21 MAY 2015 — KINGS 4, 1:30 P.M. TO 5:15 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Physiology, Behavior, and Modeling: From Middle-Ear to Mid-Brain

Sunil Puria, Chair
Stanford University, 496 Lomita Mall, Stanford, CA 94305

Contributed Papers

1:30

4pPP1. Effect of cueing on stability of behavioral measurements of basilar membrane responses with a precursor. Thibaud Necciari and Bernhard Laback (Acoust. Res. Inst., Austrian Acad. of Sci., Wohllebengasse 12-14, 1st. Fl., Vienna A-1040, Austria, thibaud.necciari@oeaw.ac.at)

Stimulation of the olivocochlear efferent system can reduce the gain of the basilar membrane (BM) response to a sound. This gain reduction may be accompanied by a reduction in BM compression, although behavioral data are inconclusive. In this study, the effect of efferent stimulation on compression is assessed using fixed-duration masking curves (FDMCs) with a precursor. In a pilot experiment, presenting a broadband noise precursor before the masker resulted in large within-listener variability for the shortest target durations, although a three-interval forced-choice task was used. To reduce possible confusion effects and increase the dynamic range of BM response measurement, a contra-lateral broadband cue signal, gated simultaneously with the precursor and masker, was added. FDMCs with a 150-ms precursor were measured with and without cue for target durations of 3, 9, and 15 ms. The total masker-target duration was fixed at 25 ms. Listeners were trained at least 2 h, then each condition was measured three times. In contrast to the literature, the presence of the cue neither reduced within-listener variability nor increased sensitivity. Alternative methods are thus required to improve the stability and dynamic range of behavioral measurements of BM responses with a precursor. [Work supported by the FWF project I-1362-N30.]

1:45

4pPP2. Distortion-product otoacoustic emission fine structure as an indicator of combination-tone-mediated masking release. James D. Lewis, Emily C. Clark, Judy G. Kopun, Walt Jesteadt, Stephen T. Neely, and Michael P. Gorga (Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, james.lewis@boystown.org)

During simultaneous masking, when the probe (fp) is higher in frequency than the masker (fm), the cubic difference tone (CDT; fcdt = 2fm-fp) can provide masking release. Given their shared origin, the 2f1-f2 distortion-product otoacoustic emission (DPOAE) may be a physiological correlate of CDT-mediated masking release [McFadden *et al.* (2012). J. Acoust. Soc. Am. **132**, 968–983]. To test this hypothesis, masking release and DPOAEs were measured in normal-hearing listeners for 2 and 4 kHz probes. Probe threshold was measured in the presence of a masker (fp/fm = 1.2) and masker-suppressor pair. The suppressor was designed to mask detection of the CDT. Masking release was calculated as the change in masked-probe threshold resulting from presentation of the suppressor. DPOAEs were measured for an f2 (f2/f1 = 1.2) between 1.8 and 2 kHz (6-Hz intervals) and between 3.6 and 4.4 kHz (12-Hz intervals). Masking release was associated with DPOAE fine structure. Listeners without masking release lacked DPOAE fine structure. Fine structure was due to wave interference between distortion- and reflection-source components; the reflection-source component is thought to originate from the place of CDT detection. Findings suggest a similar underlying cochlear mechanism for the perceptual and acoustical effects. [Work supported by NIH-NIDCD grants 1F32DC014175-01A1, 5R01DC002251-18, 2R01DC008318-06A1, and 5P30DC004662-13.]

2:00

4pPP3. Can auditory brainstem and midbrain processing of interaural level difference cues really explain perceptual performance? Stephanie T. Cheung and Ian C. Bruce (Elec. & Comput. Eng., McMaster Univ., Rm. ITB-A213, 1280 Main St. W, Hamilton, Ontario L8S 4K1, Canada, ibruce@ieee.org)

Neurons in the lateral superior olive (LSO) are believed to be involved in processing of interaural level difference (ILD) cues for sound localization. However, the ILD-tuning of LSO neurons varies with the absolute sound pressure levels (SPLs) at the two ears, in contrast to the relatively robust perceptual processing of ILD cues at different absolute SPLs. Tsai *et al*. (J. Neurophysiol. 2010) proposed that if some inferior colliculus (IC) neurons compute the difference between the contralateral and ipsilateral LSO outputs, the dependencies on absolute SPLs could be canceled out. However, they only considered pure-tone stimulation of single neurons at their characteristic frequencies and at relatively low SPLs. In this study, we evaluated the hypothesis of Tsai and colleagues using computational models of populations of auditory nerve, LSO and IC neurons consistent with Tsai and colleagues' proposal. Predictions from two different neural decoder algorithms applied to LSO and IC model outputs indicate that discharge-rate saturation and spread of excitation in the auditory nerve lead to even greater SPL-dependence of ILD coding across a population of LSO neurons than is apparent in single neurons, and the IC differencing operation proposed by Tsai and colleagues does relatively little to ameliorate this problem.

2:15

4pPP4. Effect of low-intensity highpass noise on stimulus frequency otoacoustic emission group delays for low frequency evoking tones. Jordan A. Beim and Magdalena Wojtczak (Psych., Univ. of Minnesota, N218 Elliott Hall, 75 East River Rd., Minneapolis, MN 55455, beimx004@umn.edu)

Stimulus-frequency otoacoustic emissions (SFOAEs) have been used in previous research to estimate cochlear tuning in humans. These estimates of tuning rely on the theory that SFOAEs arise from coherent reflections from the place on the basilar membrane (BM) with the characteristic frequency (CF) of the tone evoking the emission. Theories underlying SFOAE generation are still the subject of much debate, and several recent studies have shown evidence supporting an alternative theory postulating that generators of SFOAE are distributed basally to the CF place on the BM. Basally distributed emission generators could explain why SFOAE-derived group delays measured in the chinchilla cochlea for low-frequency tones were shorter than predicted by coherent-reflection hypothesis. The aim of the current study is to look for potential effects of basal emission generators in SFOAE-derived measurements of group delay at low frequencies in humans. SFOAE group delays were measured at 0.5, 0.75 kHz with and without a low intensity highpass noise, used to perturb basally distributed emission generation sites. Results show that the presence of the highpass noise leads to increased group delays estimated from SFOAEs, consistent with the hypothesis that basal generators of emission influence estimates of cochlear group delay at low frequencies.

2:30

4pPP5. The influence of short term perceptual learning of pitch discrimination and modulation discrimination on subcortical envelope-following and cortical steady-state EEG responses. Bonnie Lau, Dorea R. Ruggles, Sucharit Katyal, Stephen A. Engel, and Andrew J. Oxenham (Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455, bwlau@umn.edu)

Many perceptual skills improve with training, and research suggests that long- and short-term experiences modify auditory neural structures and function. Long-term cortical and subcortical plasticity has been associated with musical training and fluency in tonal languages, and short-term training effects have been regularly observed in cortical responses. Less is known about short-term subcortical plasticity, or the simultaneous relationships between subcortical and cortical responses under training conditions. The current study examines short-term learning and neural plasticity, using behavioral measures in combination with simultaneous subcortical and cortical steady-state EEG responses. Untrained, naïve subjects were randomly assigned to groups that were trained on fundamental-frequency (F0) discrimination, amplitude-modulation rate discrimination, or visual orientation discrimination. All auditory stimuli consisted of unresolved harmonic complexes with a nominal F0 of 137 Hz that were sinusoidally amplitude modulated at 100% depth and 13-Hz rate. Simultaneous subcortical envelope-following responses (EFR) to the F0 and cortical auditory steady-state responses (ASSR) to the modulation rate were measured pre- and post-training, and changes in the magnitude of phase locking to the respective frequency components were compared to changes in behavioral measures. The methods provide a new window on subcortical and cortical plasticity and their interactions. [Work supported by NIH grant R01DC05216.]

2:45

4pPP6. The notion of "frequency clusters" in spontaneous otoacoustic emission generation. Anthony Salerno (Medical Biophys., Univ. of Toronto, Mouse Imaging Ctr., Hospital for Sick Children, Toronto, Ontario, Canada, salerno.anthony92@gmail.com) and Christopher Bergevin (Phys. & Astronomy, York Univ., Toronto, Ontario, Canada)

Normal, healthy ears can emit unprovoked low-level sounds called spontaneous otoacoustic emissions (SOAEs). These arise in a wide variety of different species (e.g., humans, lizards) despite gross morphological differences commonly thought important to the associated biomechanics. As such, theoretical models of SOAE generation differ significantly in their underlying assumptions. One model class [Vilfan and Duke (2008), Biophys. J. **95**, 4622–4630; Wit & van Dijk, 2012 JASA **132**, 918–926] uses a coupled nonlinear oscillator framework, where each element exhibits a limit cycle (with a unique characteristic frequency) and is visco-elastically coupled to its nearest neighbor. This model proposes that SOAEs arise by groups of oscillators that self-entrain into "clusters," defined simply as the frequency where the largest peak in the steady-state spectral magnitude occurs. Our study sought to computationally explore more precisely what constitutes a cluster in terms of the underlying oscillator's dynamics. We found that oscillators within a cluster exhibit relatively complicated motions and poor phase coherence. Coupled with the model's inability to reproduce realistic SOAE spectra, the biomechanical relevance of a "cluster" is called into question. Modifications to the model for lizard ears (e.g., universal coupling via the rigid basilar membrane; Bergevin & Shera (2010), JASA 2398–2409) are explored.

3:00

4pPP7. No otoacoustic evidence for a peripheral basis underlying absolute pitch. Christopher Bergevin (Phys. & Astronomy, York Univ., 4700 Keele St., Petrie 240, Toronto, Ontario M3J 1P3, Canada, cberge@yorku.ca), Larissa McKetton (Biology, York Univ., Toronto, Ontario, Canada), Victoria Stone (Commun. Sci. and Disord., Univ. of Western ON, London, Ontario, Canada), Jessica Grahn (Psych., Univ. of Western ON, London, Ontario, Canada), and David Purcell (Commun. Sci. and Disord., Univ. of Western ON, London, Ontario, Canada)

Absolute pitch (AP) is the ability to identify or produce the perceived pitch of a sound (e.g., fundamental frequency of a piano note) without an external reference. This ability is relatively rare (~1/10,000 individuals possess it) and the mechanisms underlying AP are not well understood. This study examined whether there was evidence for a peripheral (i.e., cochlear) basis for AP based upon otoacoustic emissions (OAEs). The chief motivations were that both AP and spontaneous emissions (SOAEs) appear to have genetic components and anecdotal observations of prevalence in certain populations (e.g., relatively higher incidence of both in Asians). We examined SOAEs and stimulus-frequency emissions (SFOAEs) in both control ($N = 21$) and AP ($N = 13$) normal-hearing populations. We found no substantial differences in SOAE activity between groups (e.g., no evidence for one or more strong SOAEs that could act as a cue). SFOAE phase-gradient delays, measured using several probe levels (20–50 dB SPL), also showed no significant differences. This latter observation argues against sharper frequency selectivity in AP subjects. Taken together, these data support the prevailing view that AP mechanisms arise at a processing level in the central nervous system at the brainstem or higher (e.g., optimized neural coding).

3:15–3:30 Break

3:30

4pPP8. Nonlinear dynamics of the organ of Corti, modeling both outer hair cell somatic motility and hair bundle motility. Amir Nankali (Mech. Eng., Univ. of Michigan, 1092 Louise St., Apt. 14, Ypsilanti, MI 48197, nankali@umich.edu), Aritra Sasmal, and Karl Grosh (Mech. Eng., Univ. of Michigan, Ann arbor, MI)

One of the principal questions in the cochlea biophysics is the determination of relative contributions of the two active processes, OHC somatic motility and HB motility, to the mechanics of the cochlea. Because of the difficulty of eliminating one mechanism without affecting the other, an unambiguous *in vivo* measurement differentiating their effects remains elusive. Theoretical models, therefore, have been used as one way to examine the contributions of the two active mechanisms to cochlear mechanics. In this paper, we use a physiologically based model of the mammalian organ of Corti to study the hearing active process and the relative contributions of these active forces. This local model integrates the electrical, acoustic, and mechanical elements of a cross section of the cochlea. The nonlinear dynamics of this model are studied with a special emphasis on the regions of stability and the amplification of the mechanical response arising from the active processes. [Work supported by NIH-NIDCD R01-04084 and NIH NIDCD-T32-000011.]

3:45

4pPP9. Comparing human middle-ear motion and pressure gain across idealized and progressively more realistic three-dimensional finite-element models. Kevin N. O'Connor (Mech. Eng., Stanford Univ., 496 Lomita Mall, Durand Bldg., Rm. 283, Stanford, CA 94305, kevinoc@stanford.edu), Hongxue Cai (Mech. Eng., Stanford Univ., Chicago, IL), Peter K. Gottlieb, Charles R. Steele, and Sunil Puria (Mech. Eng., Stanford Univ., Stanford, CA)

The mammalian tympanic membrane (TM) is linked to the cochlea via a flexible and circuitous three-bone ossicular chain for which much of the mass is concentrated away from the entry axis to the cochlea. As the TM area (A_{TM}) is much larger than that of the stapes footplate (A_{FP}) and the length of the malleus (L_M) is somewhat longer than the incus (L_I), the middle ear is considered to function as a pressure transformer to optimize the flow of vibrations from low-density air to high-density cochlear fluid, with an ideal pressure gain defined as $(A_{TM}/A_{FP})*(L_M/L_I)$. Even so, the reasons for this complex ossicular arrangement, as opposed to the more straightforward case of a columella directly connecting the TM to the cochlea, are not entirely clear. We explore the effects of middle-ear anatomy and material properties on ossicular motion and pressure gain by comparing the behavior of a series of 3D finite-element models ranging from idealized simple axisymmetric cases of a flat-circular or conic TM connected to a columella, all the way to an anatomically accurate three-ossicle human middle-ear model based on a micro-CT scan of a temporal bone. In doing this, we will test whether the complex 3D anatomy of the human middle ear can be shown to offer any concrete advantages in terms of pressure gain over a simpler columella design, or whether other possible reasons for this complex design are more likely. [Work supported by R01 DC05960 from the NIDCD of the NIH.]

4:00

4pPP10. Simulating the response to clicks and the generation of spontaneous otoacoustic emissions using a cochlear model. Julien Meaud (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, julien.meaud@me.gatech.edu)

The response of the mammalian cochlea to acoustic clicks of low intensity is characterized by a long duration that includes several lobes of vibrations. Furthermore, some mammalian ears can spontaneously emit sounds, called spontaneous otoacoustic emissions (SOAEs), which are measurable in the ear canal. These phenomena are consequences of the active feedback by outer hair cells (OHCs) in the cochlea and give important information about cochlear biophysics. In this work, we use a computational model of the cochlea coupled to a lumped parameter middle ear model. Formulation of the model using a state-space approach allows determining the linear stability and the time-domain response of the model. In order to predict several lobes of vibrations in response to a click and the emission of SOAEs, inhomogeneities are introduced in the OHC properties. The model is shown to be in excellent with many aspects of the response of the cochlea to clicks that have been observed in experiments. Numerical results are used to demonstrate the strong link between characteristics of the frequency response and of the time-domain response. With small changes in the model parameters, a linearly stable model can become an oscillatory model that can emit SOAEs.

4:15

4pPP11. Computational modeling of distortion product otoacoustic emissions. Thomas Bowling, Kaikai Che, Charlsie Lemons, and Julien Meaud (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332, Tbowling3@gatech.edu)

Otoacoustic emissions are sounds generated by the cochlea that can be measured in the ear canal. Distortion product otoacoustic emissions (DPOAEs) are generated when the cochlea is stimulated by two primary tones and are a result of nonlinearities within the outer hair cells of the cochlea. DPOAEs are commonly used clinically to determine the health of the inner ear and scientifically to obtain noninvasive data about cochlear function. A physiologically based computational model of the mammalian ear is used to study the generation of DPOAEs. This model includes a nonlinear model of the cochlea, formulated in the time-domain and based on the finite element method and a lumped parameter model of the middle ear. Model simulations for the basilar membrane displacement, intracochlear fluid pressure, and ear canal pressure at the distortion product frequency in response to a two-tone stimulus are compared with experimental data and other modeling studies. The effects of various primary frequencies and input signal levels on the generation of DPOAEs are studied.

4:30

4pPP12. Short-term adaptation in acoustical and cochlear-implant hearing. Robert L. Smith (Inst. for Sensory Res., Syracuse Univ., 621 Skytop Rd., Syracuse, NY 33484, rlsmith@syr.edu) and Mahan Azadpour (Otolaryngol., NYU School of Medicine, New York, NY)

Sound onset and changes in sound intensity appear to play important roles in many aspects of hearing, e.g., detection, speech communication, and source identification. Perhaps for this reason the auditory system displays mechanisms that emphasize changes in sound intensity at multiple levels of neurophysiological processing. Short-term adaptation, observed in auditory-nerve responses, is perhaps the most peripheral example of onset emphasis. It can be observed in its simplest form in the auditory-nerve response to a constant-intensity tone burst where firing rate is maximum at response onset and decays to a steady-state value during the tone. Firing rate drops below spontaneous rate following tone offset, and the response to brief probe tones is reduced and gradually recovers in this post-stimulation interval. Short-term adaptation is generally assumed to occur in auditory hair cells and cochlear-implant stimulation by-passes hair cells and hence this potential source of adaptation. The results reviewed here will describe a model of peripheral short-term adaptation and how this model can be added to cochlear-implant speech processing. Preliminary results will be presented that indicate that adding short-term adaptation to cochlear-implant processing produces an improvement in speech intelligibility on the order of 5% to 10% in a variety of listening tasks.

4:45

4pPP13. Intra-subject variability in frequency-following responses and cortical event-related responses to Mandarin tones. Zilong Xie and Bharath Chandrasekaran (Dept. of Commun. Sci. & Disord., The Univ. of Texas at Austin, 2504A Whitis Ave. (A1100), Austin, TX 78712, xzilong@gmail.com)

Frequency-following responses (FFRs) reflect phase-locked responses from neuronal ensembles within the auditory brainstem and midbrain. Due to the high fidelity to stimulus characteristics, the FFRs have been extensively studied as biomarker, reflecting the integrity of subcortical encoding of complex auditory events. More recent studies suggest that the FFRs are highly malleable, and are influenced by higher-level attentive and cognitive mechanisms, as well as short-term auditory experiences. Here, we evaluate intra-subject variability of FFRs to complex speech sounds and examine relative

differences in intra-subject variability between FFRs and cortical evoked responses. We elicited FFRs to two Mandarin tones (tone 1 and tone 2) from four native Mandarin speakers over five repeated daily sessions within a week. We also recorded cortical evoked responses to these two Mandarin tones presented in an oddball paradigm from these participants during these five sessions. Several measures were used to quantify FFRs: signal-to-noise ratio, stimulus-to-response correlation, and response- to-response correlation. Our analyses show that FFRs exhibited low intra- subject variability, whereas cortical responses such as N1, P2, and mismatch-negativity demonstrated higher intra-subject variability. Our findings demonstrate high individual stability in the FFRs and FFRs can be reliable biomarker to study the integrity of subcortical encoding of speech sounds.

5:00

4pPP14. Variability in word recognition by adults with cochlear implants: The roles of perceptual attention versus auditory sensitivity. Aaron C. Moberly (Otolaryngol., The Ohio State Univ., 915 Olentangy River Rd., Ste. 4000, Columbus, OH 43212, Aaron.Moberly@osumc.edu)

Enormous outcome variability exists for postlingually deafened adults who receive cochlear implants (CIs). This variability is not purely a result of the degradation of speech representations through implants. Data from two studies are presented that examined variability in word recognition as explained by "perceptual attention" and "auditory sensitivity" to acoustic cues underlying speech perception. In each experiment, postlingually deafened adults with CIs and normal-hearing (NH) controls performed three tasks: (1) word recognition in quiet; (2) labeling for three sets of stimuli that varied based on acoustic cues in the amplitude, duration, or spectral domains; and (3) discrimination of nonspeech analogs of these acoustic cues. Word recognition was found to vary widely among CI users (20 to 96%). Attention to spectral cues, as measured by weighting coefficients computed from labeling functions, beyond simple sensitivity to spectral cues, as measured by d' values, predicted variability in word recognition. Attention and sensitivity to nonspectral cues did not predict variability in word recognition. Efforts should be made to better represent spectral cues through implants; however, facilitating attention to these cues through auditory training should also be a clinical focus.

THURSDAY AFTERNOON, 21 MAY 2015 KINGS 3, 2:00 P.M. TO 4:05 P.M.

Session 4pSA

Structural Acoustics and Vibration, Education in Acoustics, and Physical Acoustics: Demonstrations of Structural Acoustics and Vibration

Daniel A. Russell, Chair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Chair's Introduction—2:00

Invited Papers

2:05

4pSA1. The cymbal as a structural acoustics demonstration: From evanescence to chaos. Kent L. Gee, Scott D. Sommerfeldt, Trevor A. Stout, Tracianne B. Neilsen, and Pegah Aslani (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

The cymbal can be used as a rich structural acoustics demonstration. Many well-defined normal modes spanning a broad range of frequencies can be evaluated using various techniques, including Chladni patterns, speckle interferometry, Doppler vibrometry, and acoustical holography. The literature shows modal patterns that are both membrane-like with nodal lines and circles as well as more complicated, irregular shapes. When the cymbal is struck forcefully, orthogonality gives way to modal coupling and resulting chaotic vibration that produces the characteristic shimmering sound. The sound radiation favors high-frequency modes; however, when the cymbal is put close to the ear, low frequencies dominate the sound heard. This low-frequency ringing persists well after the rest of the sound has decayed. This auralized evanescence will be demonstrated and further described as part of the presentation.

2:25

4pSA2. Demonstrations of fundamental topics in structural acoustics using a simply-supported plate. Andrew R. Barnard (Mech. Eng. - Eng. Mech., Michigan Technolog. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, arbarnar@mtu.edu) and Stephen A. Hambric (Graduate Program in Acoust., Penn State Univ., State College, PA)

As part of the ACS 519 course, sound-structure interaction, offered through the Penn State Graduate Program in Acoustics a set of demonstrations was developed to help reinforce structural acoustics theory. A simply-supported plate was used to develop a set of four concept demonstrations: mobility functions, radiated sound directivity, fluid loading, and acoustic transmission loss. A modal impact hammer, accelerometers, microphones, a sound level meter, and a sound intensity probe were used in combination with National

Instruments compact DAQ systems and LabVIEW software to develop these custom demonstrations. Basic theory and setup of the demonstrations are presented as well as videos of the demonstrations themselves.

2:45

4pSA3. A glass half full: Demonstrations of some surprising effects of fluid loading on the resonance of wine glasses (and other vessels). Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

A popular and (perhaps) provocative demonstration in an introductory course on acoustics and vibration involves asking students to predict what happens to the tap tone of a wine glass when liquid is added. Although their predictions are often incorrect, students readily accept the explanation of mass loading for why the pitch decreases, having already seen what happens to the frequency of a mass-spring system when mass is added. But is the pitch-lowering effect the same for all vibrational modes? Does it depend on the shape or thickness of the glass? For students beyond the introductory level, these questions stimulate deeper thinking about fluid-structure interaction. Predictions can be verified with a very simple apparatus.

3:05

4pSA4. Vibrational modes of a thin bar. Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Appl. Res. Lab, P. O. Box 30, State College, PA 16804, sxg185@psu.edu)

Although most traditional acoustics textbooks that treat vibration (e.g., Rayleigh, Lamb, Morse, Morse and Ingard, Kinsler and Frey) address the longitudinal, torsional, and flexural modes of a thin bar, they do not stress the importance of sound speed measurements for the experimental determination of elastic moduli. This presentation will demonstrate a simple and inexpensive apparatus that uses two coils of copper wire and permanent magnets, developed by Bob Leonard and Izzy Rudnick at UCLA, that can be used to excite and detect all three modes of a free-free bar, accurately measure their modal frequencies, and use those results to provide accurate and precise values for the elastic moduli of the bar material [JASA **88**(1), 210 (1990)].

3:25

4pSA5. Standing waves on an electrically heated wire, revisited: Demonstration of glow at the nodes. Murray S. Korman and Ted McClanahan (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

In the experiment, a nichrome wire is suspended between a metal bridge and a pulley with a weight hanger and standard weights that supply a constant tension. The 28 gauge wire has a diam = 0.32 mm and a mass/length = 0.66 g/m. For a 2 m long wire with a 0.43 kg mass providing tension, the standing wave resonant frequencies are integer multiples of 20 Hz (neglecting the wire's stiffness). An AC current from a variable autotransformer is used to heat the wire (from the bridge to the pulley) until it glows. Near the bridge, a variable gap neodymium magnet is positioned with its poles on either side of the wire. The wire will glow at the nodes, but due to cooling (by the air vibration) the antinodes do not glow. The experiment is described in [C. A. Taylor, *The Art and Science of Lecture Demonstration*, Taylor & Francis, 1988]. Replace the magnet with an oscillator and audio power amplifier for frequency control [Sabatier and Drewrey, J. Acoust. Soc. Am. **104**, 1793 (1998)]. One can also demonstrate glowing nodes for "standing waves in a circle," by modifying the setup of [H. F. Meiners, Am. J. Phys. **33**, xiv, Oct. 1965].

3:45

4pSA6. Guitar pickups and the plucked string. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, drussell@engr.psu.edu)

This demonstration will use optical, magnetic, and piezoelectric guitar pickups to measure time signals and frequency spectra for the displacement, velocity, and force (measured at the bridge) of a plucked and struck guitar string. For the plucked string, experimental observations of the displacement and velocity of a point on the string agree very well with theory. Frequency spectra also agree well with theoretical predictions and highlight the differences in timbre between the audio signals obtained with each pickup. The demonstration will briefly discuss theoretical time signals and frequency spectra for a point on a plucked string, show measured time signals and frequency spectra from the various pickups, and compare audible examples of the sounds from the pickups for a plucked string. For the struck string, a simple theory approximates the basic shape of the string displacement and velocity, but dispersion effects in the string and the reality of the hammer-string interaction are evident in the measured string response. If time permits, a demonstration of the sound and spectrum of a compound string will be included.

Session 4pSC

Speech Communication: Indexical Factors in Speech Perception and Production (Poster Session)

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

Contributed Papers

4pSC1. Acoustic effects of loud speech across vowels and speakers. Laura L. Koenig (Haskins Labs and Long Island Univ., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu) and Susanne Fuchs (Zentrum für Allgemeine Sprachwissenschaft [ZAS], Berlin, Germany)

Although loud speech is widely agreed mainly to involve increased respiratory driving support leading to greater acoustic intensity, past work suggests that speakers may make laryngeal and supralaryngeal adjustments as well. For example, previous studies suggest that loud speech is accompanied by greater jaw displacements as well as formant frequency variation, especially in the form of higher first formants. Acoustic studies have generally not sampled widely across the vowel space, however. This study assessed how formants, f0, duration, and intensity varied across conversational and loud conditions, with a particular focus on formant frequencies. Eleven German-speaking women performed three speaking tasks: Reading, answering questions, and recounting a recipe. Target words included a range of high, low, tense, and lax vowels. Loudness variation was elicited naturalistically via changing interlocutor distance. Initial results from the reading and question-answer tasks suggest that formant frequencies vary on average in louder speech, particularly for F1, but the effects are speaker-dependent and may also differ between high vs. low and tense vs. lax vowels. These acoustic data will ultimately be combined with simultaneously recorded data on intraoral pressure, vocal-fold contact, and breathing kinematics to assess the range of speakers' strategies for achieving louder speech.

4pSC2. Interactions among phonetic reduction and sociolinguistic variation in word recognition. Zack E. Jones and Cynthia G. Clopper (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, jones.5028@osu.edu)

A number of linguistic factors influence the phonetic realization of segments within a word, especially vowels within the word. These factors include lexical frequency, lexical neighborhood density, semantic predictability, and mention within a discourse. Sociolinguistic factors, such as speaking style and regional dialect, are also known to influence the phonetic realization of vowels. This study investigates the ways in which phonetic reduction caused by the linguistic factors and sociolinguistic variation from speaking style and talker dialect affect lexical processing. Participants are asked to identify single word tokens in noise that vary in frequency, density, predictability, discourse mention, speaking style, and talker dialect in a fully crossed design. We expect participants to identify words containing vowels with less phonetic reduction, a clear speaking style, and a familiar regional dialect with greater accuracy than words with more reduced vowels, a plain speaking style, or a less familiar dialect. The experimental design allows us to explore the individual contributions to word intelligibility of each of the linguistic and sociolinguistic factors independently and in complex interactions.

4pSC3. Talker normalization and acoustic properties both Influence spectral contrast effects in speech perception. Ashley Assgari and Christian Stilp (Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, ashley.assgari@louisville.edu)

Talker normalization (TN) occurs when listeners adjust to a talker's voice, resulting in faster and/or more accurate speech recognition. Several have suggested that TN contributes to spectral contrast, the perceptual magnification of changing acoustic properties. Studies using sine tones in place of speech demonstrated that talker information is not necessary to produce spectral contrast effects (Laing *et al.*, 2012 *Front. Psychol.*). However, sine tones lack acoustic complexity and ecological validity, failing to address whether TN influences spectral contrast in speech. Here we investigated how talker and acoustic variability influence contrast effects. Listeners heard sentences from a single talker (1 sentence), HINT (1 talker, 200 sentences) or TIMIT databases (200 talkers, 200 sentences) followed by the target vowel (varying from "ih" to "eh" in F_1). Low (100–400 Hz) or high (550–850 Hz) frequency regions were amplified (+20, +5 dB) to encourage "eh" or "ih" responses, respectively. When sentences contained +20 dB spectral peaks, contrast effect magnitudes were comparable across conditions. When sentences contained +5 dB peaks, contrast effect magnitudes decreased overall, but were smallest following TIMIT sentences with larger, comparable effects for single-talker conditions. Thus, TN influences contrast effects when spectral peaks are modest, but not when they are large.

4pSC4. Speaker age effects on the voicing contrast of Tokyo Japanese stops. Zack E. Jones and Chris S. Rourke (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, jones.5028@osu.edu)

While older Japanese speakers from the Kansai region produce voiced-voiceless stop contrasts with a true voicing distinction, older speakers from the northern Tohoku region produce the same stop contrasts with a short- versus long-lag VOT distinction, similar to the voicing contrast of English stops. However, Japanese speakers of younger generations from both of these dialect regions have been observed to produce the voiced-voiceless stop contrast in much the same way to each other, and they also seem to be using VOT as a less informative cue for stop distinction than speakers of previous generations (Takada, 2011). This study investigates the status of this sound change in the speech from the Tokyo dialect region, generally considered the standard for the modern language. Using speech data available from the Corpus of Spontaneous Japanese, we analyze stops produced by Tokyo speakers to determine differences in voicing contrast conditioned by speaker age and gender. We expect the stop contrast of older speakers to pattern differently than that of younger speakers, and we expect to see evidence for a sound change in progress similar to that observed in the other dialects. We also explore other acoustic correlates that may potentially contribute to Japanese stop contrasts as the sound change progresses.

4pSC5. Regional variation of vowels in Saterland Frisian. Heike Schoormann, Wilbert Heeringa, and Jörg Peters (Inst. of German Studies, Carl von Ossietzky Univ. Oldenburg, Ammerlaender Heerstraße 114-118, Oldenburg 26129, Germany, heike.schoormann@uni-oldenburg.de)

Saterland Frisian is spoken in three neighboring villages in North-West Germany, Strücklingen, Scharrel, and Ramsloh. In this study, we examine whether there is regional variation between the vowel systems of the three varieties. Speakers were instructed to read monophthongs and diphthongs in a neutral /hVt/ frame. Acoustic measurements included vowel duration, mid-vowel F1 and F2, the amount of Vowel Inherent Spectral Change (VISC, Nearey and Assmann, 1986), and the spectral rate of change (cf. Fox and Jacewicz, 2009*)*. Results confirm large inventories for the three varieties of Saterland Frisian, although some vowels have undergone a merger with neighboring categories. The comparison of single vowel categories in the three varieties did neither reveal variation of vowel duration nor of dynamic spectral properties. Regarding static spectral properties, however, dialectal variation was observable. In Scharrel, monophthongs are more centralized in the F1 dimension. This finding is discussed with respect to the common view of local people that speakers from Scharrel speak faster than speakers from the other two places.

4pSC6. Vowels of Korean dialects. Yoonjung Kang (Dept. of Linguist, Univ. of Toronto, 1265 Military Trail, Humanities Wing, H427, Toronto, Ontario M1C 1A4, Canada, yoonjung.kang@utoronto.ca), Jessamyn L. Schertz (Ctr. for French and Linguist, Univ. of Toronto Scarborough, Toronto, Ontario, Canada), and Sungwoo Han (Korean Lang. and Lit., Inha Univ., Seoul, South Korea)

This study compares the monophthongal vowels /a ɛ e i ʌ ɨ o u/ of two North Korean dialects as spoken by ethnic Koreans in China (24 Phyeongan and 21 Hamkyoung) with the vowels of Seoul Korean (25 younger and 32 older). Younger and older speakers of Seoul Korean are compared to examine the sound change in progress in Seoul. The most striking difference among the dialects is in the realization of /o/ and /ʌ/. In Seoul, /o/ is produced higher than /ʌ/. In Phyeongan, /o/ is lower than /ʌ/, while in Hamkyoung, the two are comparable in height and the main contrast is along F2. Also, /e/-/ɛ/ contrast is lost in Seoul but robust in the Northern dialects. Within Seoul Korean, the back vowel shift observed in recent literature is confirmed (Cho S. 2003, Han J. and Kang H. 2013, and Kang Y. to appear)—/o/ is raised toward /u/ while /ɨ/ is fronted away from /u/ in younger speakers' speech. In contrast to recent reports of /u/-/ɨ/ and /o/-/ʌ/ merger in homeland North Korean dialects (Kang S. 1996, 1997, Kwak 2003, and So 2010), in our Northern data, these contrasts remain distinct.

4pSC7. Diachronic change in perception of Korean sibilants. Jessamyn L. Schertz, Yoonjung Kang (Ctr. for French and Linguist, Univ. of Toronto Scarborough, 1265 Military Trail, Humanities Wing, HW427, Toronto, Ontario M1C 1A4, Canada, jessamyn.schertz@utoronto.ca), Sungwoo Han (Korean Lang. and Lit., Inha Univ., Seoul, South Korea), and Eunjong Kong (Dept. of English, Korea Aerosp. Univ., Seoul, South Korea)

The laryngeal contrast in the Seoul dialect of Korean is in a state of flux: the increasing importance of f0 (relative to VOT) in both perception and production of the three-way stop contrast in younger (vs. older) Seoul speakers has been well-documented. The current work turns to perception of the Korean sibilant series, comprised of a three-way affricate contrast, (fortis vs. lenis vs. aspirated, parallel to the stop contrast) and a phonologically ambiguous two-way fricative contrast (fortis vs. "nonfortis"). We map younger (mean 33 years old) and older (mean 66) Seoul listeners' perceptual spaces for the sibilant class using a five-way forced-choice task, with stimuli manipulated to vary independently across multiple acoustic dimensions (consonantal spectral information, vocalic spectral information, frication duration, aspiration duration, and f0). Hierarchical classification tree analyses reveal systematic age-related differences in cue-weighting. While both age groups make use of a combination of spectral properties of both the consonant and vowel, temporal information, and f0 when categorizing the stimuli, f0 plays a greater role in predicting sibilant classification in younger as compared to older listeners. Furthermore, categorization patterns suggest that the sound change not only affects perception of the three-way laryngeal contrast, but also has implications for other phonological contrasts (e.g. affricate vs. fricative "manner" contrast).

4pSC8. The southern vowel shift in women from Mississippi. Whitney Knight and Wendy Herd (English, MS State Univ., 2004 Lee Hall, MS State, MS 39762, whitneyleighknight@gmail.com)

Though previous research has documented the Southern Vowel Shift (SVS) in Alabama and Tennessee, no research has focused on the SVS in Mississippi. The majority of SVS research has also focused on European-Americans and assumed that African-Americans do not participate in the shift. The SVS consists of three stages: /aɪ/ monophthongization; lowering and centralizing of /e/ toward /ɛ/ and raising and peripheralizing of /ɛ/ toward /e/; and lowering and centalizing of /i/ toward /ɪ/ and raising and peripheralizing of /ɪ/ toward /i/. In this study, data were collected from women from northern ($N = 11$) and central ($N = 23$) Mississippi, with central residents evenly recruited from urban and rural areas. Of these, 15 were European-American and 19 were African-American. Participants read a list of words including the target vowels in b_t and b_d frames, and then F1 and F2 were measured at five equidistant points. F1, F2, and trajectory length were analyzed to determine to what extent participants exhibited the SVS. There were effects of Region, Rurality, and Race such that central residents shifted more than northern residents, rural residents shifted more than urban residents, and African-American residents shifted more than European-American residents. These results suggest that African-Americans do participate in the SVS.

4pSC9. /l/-darkness in Newfoundland English. Sara Mackenzie, Paul De Decker, and Rosanna Pierson (Linguist, Memorial Univ. of NF, Sci. Bldg. 3050B, St. John's, New Foundland A1B 3X9, Canada, paulde@mun.ca)

This paper reports on the first acoustic study on the allophonic distribution of /l/ in Newfoundland, Canada, where Irish-settled varieties of English exhibit light variants in both coda and onset positions (Clarke 2010, Paddock 1982). This pattern is distinct from standard North American English which has a dark, or velarized form in coda position and light [l] in onsets (Halle & Mohanan 1985). Productions from 10 male and 13 female speakers from across the province were collected and spectrally analyzed using Praat 5.4 (Boersma & Weenink 2014). Darkness measurements (F2-F1 at the midpoint of each /l/) were calculated and within-subjects t-tests conducted to examine differences across onset and coda positions. Results show the standard North American pattern exists within our sample, with significantly darker /l/s in coda position. However, some speakers showed coda darkness values comparable to onset /l/s of those speakers with the allophonic variation. Others had darkness values comparable to coda realizations in all positions. These findings are discussed using a model of dialect contact that identifies how, despite the presence of the dominant North American pattern, some speakers of Newfoundland English maintain a several hundreds year old pattern which lacks allophonic variation across syllable positions.

4pSC10. Production and perception among three competing pre-/l/ mergers. Lacey R. Arnold (English, North Carolina State Univ., 1214 Carlton Ave., Raleigh, NC 27606, lrarnold@ncsu.edu)

This study examines the status of three patterns of merger among /ul/, /ol/, and /ʊl/ in Youngstown, Ohio. Using acoustic analyses of F1 and F2 and multiple-forced-choice perception task results from 40 Youngstown natives ages 9–81, this study addresses the progression of these mergers in apparent time, the perceptual correlates of merged and distinct production, and maintenance of durational distinctions in production and/or perception. Initial analysis of perception data suggests that different patterns of merger are progressing differently in the community and that production does not directly correlate with perception, perhaps as a result of exposure to multiple patterns of merger in the community. Despite the different patterns of merger in the community, speakers seem to pick up on durational cues in perception tasks.

4pSC11. Dialectal duration variations reveal historical sound change. Di Wu and Chilin Shih (Linguist, Univ. of Illinois at Urbana-Champaign, 4080 Foreign Lang. Bldg., 707 S Mathews Ave., Urbana, IL 61801, diwu4@illinois.edu)

This study models duration patterns of different dialectal regions to evaluate the spread of a phonological process. Er [ɚ] is a diminutive suffix in Chinese and is a major linguistic feature differentiating northern and southern Mandarin speech. This study evaluates the duration of the er-suffix along with its host syllable in a sentence reading task. The participants were from three regional groups chosen to examine the spread of this feature—Beijing, Mid-China, and Taiwan. Linear regression models were used to analyze duration variations quantitatively. Results reveal that Beijing speakers exhibit the most advanced er-suffixation, as expected. Duration values of words with or without the er-suffix are nearly identical, suggesting that er has been incorporated into the host syllable and has lost its own syllabic status. In contrast, speakers from Taiwan show the least advanced er-suffixation, where the suffix maintains its syllabic status with full duration. Speakers from Mid-China show an intermediate stage in accord with the geographical location. The duration contrast is consistent with reported diachronic phonological changes. The method suggests that duration patterns can be used to evaluate the spread of phonological processes, projecting synchronic, regional variations to their diachronic development.

4pSC12. Talker variation in the perception of speech in noise. Noah H. Silbert and Lina Motlagh Zadeh (Commun. Sci. & Disord., Univ. of Cincinnati, 3202 Eden Ave., 344 French East Bldg., Cincinnati, OH 45267, noah.silbert@uc.edu)

Speech communication commonly occurs in the presence of noise. Research on the perception of speech in noise has largely focused on the perceptual availability of target phonetic information in the presence of various noise maskers (e.g., white noise, speech-shaped noise, temporally modulated noise, and multi-talker babble). Previous work has shown that a glimpsing model and a modified Articular Index model closely approximate overall accuracy of noise-masked speech perception (Cooke, 2006, *JASA*; Allen, 2005; *JASA*). Some recent work has focused on variation in confusion patterns across listeners (Silbert, 2012, *JASA*, 2014, *Lab. Phon.*) and on difference in perceptual error rates across and within consonant categories (Toscano & Allen, forthcoming, *JSLHR*). The present work focuses on listener and stimulus-talker variation in the identification of consonants. Eleven listeners identified numerous tokens of each of four consonants (t, d, s, z) in CV syllables produced by 20 talkers (10 male, 10 female) masked by 10-talker babble. Data plots and multilevel logistic regression model fits indicate substantial variation in talker-specific influences on identification accuracy as well as substantial interactions between talker and stimulus category. Analysis of perceptual confusion patterns via a multilevel, multidimensional Gaussian signal detection model will also be presented.

4pSC13. Human recognition of familiar voices. Stanley Wenndt (Air Force Res. Lab., 701 N George St., Rome, NY 13440, swenndt@twcny.rr.com)

Recognizing familiar voices such as friends and co-workers is something we do every day. In typical environments such as an office or home, it is usually easy to identify who is speaking. In noisier environments, this can become a difficult task. This research examines how robust listeners are at identifying familiar voices in noisy, changing environments and what factors may affect their recognition rates. While there is previous research addressing familiar speaker recognition, it is a difficult topic to research since the focus is on familiar voices. The data being used for this research was collected in such a fashion to mimic conversation, free-flow dialog, but in a way to eliminate many variables such as word choice, intonation or non-verbal cues. This data provides some of the most realistic test scenario to-date for familiar speaker identification. A pure-tone hearing test was used to separate speakers into normal hearing and hearing impaired groups. It is hypothesized that the results of the Normal Hearing Group will be statistically better than then results of the Hearing Impaired Group. Additionally, a new aspect of familiar speaker recognition is addressed by having each listener rate his or her familiarity with each speaker.

4pSC14. Sensitivity to the emotional tone of verbal commands: Executive-function and chemotherapy effects. Blas Espinoza-Varas, Sudha Lakhwani (Commun. Sci. & Disord., OU Health Sci. Ctr., 1200 N. Stonewall Ave., Oklahoma City, OK 73117-1215, blas-espinoza-varas@ouhsc.edu), Ashan Khan (IPS Researrch, Oklahoma City, OK), and Kai Ding (Biostatistice & Epidemiology, OU Health Sci. Ctr., Oklahoma City, OK)

Treatment adherence requires complying with verbal commands ("quit smoking!") issued to impede or instigate behaviors, and a lenient or stern emotional voice tone (VT) calls for optional or mandatory adherence. The ability to identify the commands VT in conditions imposing executive-function (EF) demands could estimate the likelihood of achieving adherence. Prior to chemotherapy and between cycles 3–4 of gynecological cancer, we assessed VT identification under three EF demands. 1) Inhibitory control trials presented the cue word "left" or "right" followed by impeding commands in lenient or stern tone, mapped onto a left or right response; the cue and ear side could be congruent or in conflict with the correct response side. Trials presenting instigating commands ("go!") mapped lenient or stern onto a right or left response. 2) Response-mapping switching conditions interleaved impeding and instigating commands within the same trial block, and required switching the mapping rule depending on the command, impeding or instigating. 3) Working-memory conditions asked whether the command presented on the current trial was equal to or different from the one presented two trials back. Without EF demands, VT identification errors were few, but increased significantly with EF demands, being largest in condition 2; chemotherapy effects were small.

4pSC15. A language familiarity effect for talker identification in forward but not time-reversed speech. Sara C. Dougherty, Deirdre E. Mclaughlin, and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 653 Commonwealth Ave., Boston, MA 02215, sarad12@bu.edu)

Listeners identify voices more accurately in their native language than an unfamiliar foreign language—a phenomenon known as the Language Familiarity Effect (LFE). The purpose of this study was to assess two hypotheses about the source of the LFE: 1) that the LFE depends on linguistic processing of speech, or 2) that it results from listeners' familiarity with language-specific but non-linguistic properties of speech acoustics. We trained native speakers of English ($N = 16$) and Mandarin ($N = 16$) to identify English and Mandarin voices. Stimuli consisted of unaltered recordings of ten sentences in those languages, as well as time-reversed versions of the same. Importantly, time-reversed speech preserves global acoustic properties of the stimuli while rendering them completely incomprehensible. Consistent with the LFE, participants identified voices in their native-language significantly more accurately than in the foreign language when listening to normal, forward speech. However, participants did not exhibit a corresponding LFE for the time-reversed speech stimuli: time-reversed native- and foreign-language voices were identified with equal accuracy. These findings support the hypothesis that processes involved in speech comprehension are the source of the LFE rather than differences in the non-linguistic analysis of speech acoustics.

4pSC16. Effects of reading ability on native and nonnative talker recognition. Minal Kadam (Univ. of Connecticut, Storrs, CT), Adriel J. Orena (McGill Univ., Montreal, Quebec, Canada), Rachel M. Theodore (Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu), and Linda Polka (McGill Univ., Montreal, Quebec, Canada)

Recent findings suggest that phonological knowledge influences talker identification. Specifically, talker identification is improved for native compared to nonnative talkers, and adults with reading disability show impaired talker identification even for native talkers [Perrachione *et al.*, Science, **333,** 595 (2011)]. Here, we examine whether effects of reading ability on talker identification emerge among unimpaired readers. Monolingual English adults were assigned to either the high or low reading group based on standardized assessments of reading and reading sub-skills. All readers learned to identify the voices of four English talkers and four French talkers. Training consisted of a two-alternative forced choice task with feedback provided on every trial. After training, retention of learning was tested using a four-

alternative forced choice task without feedback. The results to date suggest that the high reading group learned both the native and nonnative voices faster compared to the low reading group. Moreover, the high reading group showed increased retention of learning compared to the low reading group, but only for the nonnative voices. These results are consistent with recent findings demonstrating an effect of language proficiency on talker identification, and extend them to include a gradient role for native language phonological ability on nonnative talker identification.

4pSC17. Cross-modal transfer of talker learning. Dominique C. Simmons, James W. Dias, Josh Dorsi, and Lawrence D. Rosenblum (Psych., Univ. of California Riverside, 900 University Ave., Riverside, CA 92521, dsimm002@ucr.edu)

Observers can match unfamiliar faces to corresponding voices (Kamachi *et al*., 2003). Observers can also use dynamic point-light displays containing isolated visible articulations to match faces to voices (Rosenblum *et al*., 2006) and to sinewave versions of those voices (Lachs & Pisoni, 2004) suggesting that isolated idiolectic information can support this skill. Cross-modal skills also extend to facilitation of speech perception. Familiarity with a talker in one modality can facilitate speech perception in another modality (Rosenblum, Miller, & Sanchez, 2007; Sanchez, Dias, & Rosenblum, 2013). Using point-light and sinewave techniques, we tested whether *talker* learning transfers across modalities. If learning of idiolectic talker information can transfer across modalities, observers should better learn to auditorily recognize talkers they have previously seen. Sixteen subjects trained to recognize five point-light talkers. Eight of these subjects then trained to recognize sinewave voices of the five previously seen talkers (Sheffert *et al*., 2002). The remaining subjects trained to recognize five sinewave voices of new talkers. Subjects trained to recognize voices of talkers who they had previously seen performed better than subjects trained to learn voices of new talkers, $t(14)=-1.834$, $p<0.05$. Results suggest that learning idiolectic talker-specific information can transfer across modalities to facilitate talker learning.

4pSC18. Exposure to an unfamiliar language bolsters talker learning. Adriel John Orena (School of Commun. Sci. and Disord., McGill Univ., 2001 McGill College, 8th Fl., Montréal, Quebec H3A 1G1, Canada, adriel.orena@mail.mcgill.ca), Rachel M. Theodore (Dept. of Speech, Lang., and Hearing Sci., Univ. of Connecticut, Storrs, CT), and Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montréal, Quebec, Canada)

Listeners are better at identifying talkers who speak their native language than talkers who speak a foreign language, showing that phonological knowledge of a language facilitates talker identification. However, research with infants indicates that language comprehension is not necessary for improving talker identification. In this study, we asked whether language exposure alone could improve talker learning. Two groups of English-monolingual adults were recruited: one group from Montréal, Québec, who receive regular French exposure, and the other from Storrs, Connecticut who receive no French exposure. In Experiment 1, we used a four-alternative forced choice task (4AFC) to train listeners about the voices of four English talkers and four French talkers. Results show that Montréal participants were faster at learning French voices than Storrs participants, showing that exposure to a foreign language is sufficient in boosting talker learning in that language. However, in Experiment 2, a 2AFC was used to train participants, and no significant differences were found between groups. These findings show that varying the training paradigm in laboratory analogs of talker identification can induce different types of talker learning. Taken together, our results suggest that phonological sensitivity contributes to listeners' talker identification abilities, but only under certain training contexts.

4pSC19. Processing talker variability in semantic/associative priming: Does talker voice matter? Yu Zhang and Chao-Yang Lee (Ohio Univ., Grover Ctr. W239, Athens, OH 45701, yz137808@ohio.edu)

The effect of talker variability on lexical access is investigated using short-term semantic/associative priming experiments. Prime-target pairs, either semantically associated (e.g., king-queen) or unrelated (e.g., bell-queen), were spoken by the same or different male speakers. Two interstimulus intervals (ISI, 50 and 250 ms) were used to explore the time course of semantic/associative priming and voice specificity effects. Forty listeners completed a lexical decision task followed by a talker voice discrimination task on the same auditory stimuli. Results from the lexical decision task showed semantic/associative priming effects, although the magnitude of priming was unaffected by talker variability or ISI. Results from the talker voice discrimination task showed no priming effects, although different-talker trials elicited faster and more accurate responses. In comparison with previous results using a similar paradigm (Lee & Zhang, in press; Zhang & Lee, 2011), this set of data suggests that talker variability might not influence access to semantic aspects of spoken language. Furthermore, voice discrimination task generated different patterns (e.g., lack of priming) from the lexical decision task. This suggests that the extent of talker voice effect on lexical access may be subject to attention manipulation.

4pSC20. Gender and age differences in vowel-related formant patterns: What happens if men, women, and children produce vowels on different *and* on similar F0? Dieter Maurer, Heidy Suter (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Toni-Areal, Pfingstweidstrasse 96, Zurich 8031, Switzerland, dieter.maurer@zhdk.ch), Daniel Friedrichs, and Volker Dellwo (Dept. of Comparative Linguist, Univ. of Zurich, Zurich, Switzerland)

There is a broad consensus in the literature that vowel-specific formant patterns differ as a function of gender (men/women) or age (adults/children) due to different average vocal tract sizes. Although an additional influence of fundamental frequency F0 is discussed in corresponding normalization approaches, formant patterns relating to sounds of adults and children that exhibit the same F0, to sounds of adults with higher F0 than sounds of children, and to sounds of men with higher F0 than sounds of women are barely compared. Investigating vowels of men, women, and children producing sounds with varying F0, we observed (1) a possible decrease or even a disappearance of the expected speaker-group differences in the formant frequencies < 1.5 kHz if F0 of the utterances correspond for children, women, and men, and (2) a possible "inversion" of the expected speaker-group differences < 1.5 kHz if F0 of the utterances of adults are higher than those of children, or F0 of men are higher than those of women. However, no corresponding relationship between F0 and the higher formants > 1.5 kHz was found. These observations call for a further examination of the role of F0 when interpreting speaker-group related differences in formant patterns.

4pSC21. Speaker sex discrimination performance for voiced and whispered vowels at very short durations. David R. Smith (Psych., Univ. of Hull, Cottingham Rd., Hull HU6 7RX, United Kingdom, d.r.smith@hull.ac.uk)

When listening to someone's voice what duration of stimulus is required to tell whether the person speaking is a man or a woman? Previous research studied this using *voiced* speech [D. R. R. Smith, Acta Psychologica **148**, 81–90 (2014)]. The current study expanded this analysis by investigating what duration of stimulus is required to discriminate speaker sex when listening to *whispered* speech. Psychometric functions were collected plotting percent correct discrimination that a man or woman spoke, as a function of very brief vowel segment durations (up to 60 ms), for both voiced and whispered vowels. Results show that speaker sex discrimination performance is significantly impaired for whispered vowels, as compared to voiced vowels, for all durations tested. These findings are interpreted in terms of the impoverished cues to speaker sex in whispered compared to voiced speech.

4pSC22. Gender normalization in fricative perception in single- and mixed-gender blocks. Benjamin Munson (Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455, benjamin.ray.munson.jr@gmail.com)

Previous research has shown that listeners' identification of English anterior sibilant fricatives changes depending on whether they are primed to believe that the talker is a woman or a man, by pairing audio stimuli with images or videos of a woman or a man (Munson, 2011; Strand & Johnson, 1996; Winn *et al*., 2013). The current experiment is part of a program of research that endeavors to understand the nature of imputed gender effects in speech perception. Previous studies used fully within-subjects designs: listeners were presented with all stimuli combined with both male and female faces. The current experiment examined whether gender normalization occurs equally strongly in within-groups designs, and between-groups designs in which listeners are presented with only male or female faces. Four groups of listeners identified a nine-step *sack-shack* continuum created by combining a s-ʃ continuum with a natural production of a VC that had been acoustically manipulated to be gender-neutral. Listeners were presented with a male face only, a female face only, both male and female faces in separate trials, or no face. Results will indicate how robust gender normalization effects are across different experimental conditions.

4pSC23. Identification of talker gender and speech production mode from high-pass filtered vowel segments. Jeremy Donai (Commun. Sci. and Disord., West Virginia Univ., 3601 4th St., Lubbock, TX 79430, jeremy.donai@ttuhsc.edu) and Dwayne Paschall (Speech-Lang-Hearing Sci., Texas Tech Univ. Health Sci. Ctr., Lubbock, TX)

This study reports gender and production mode judgments from high-pass filtered vowel segments from 22 listeners with normal-hearing. Two adult males and two adult females produced vowels in an h/Vowel/d (hVd) context in both a spoken and sung mode of production. The hVd utterances were produced using the carrier phrase, "I say (insert hVd) again." The speakers produced the sung utterances to the tune of "Old McDonald." The vowels included /i/ as in "heed", /ɝ/ as in "heard", and /ɔ/ as in "hawd." A 250 millisecond segment from the central portion was extracted and windowed. The signals were then high-pass filtered at 3.2 kHz to remove low-frequency spectral detail. The listeners were positioned in front of a graphical user interface and instructed to click on one of four buttons, "Male spoken," "Male sung," "Female spoken," or "Female sung" to indicate the perceived gender and mode of production. Results showed below chance performance for the male sung and female spoken utterances and performance significantly above chance for the male spoken and female sung utterance. Signals with similar unfiltered F0 values (i.e., male sung and female spoken) were discriminated compared to those with the highest (female sung) and lowest (male spoken) F0s.

4pSC24. Vowels or consonants: Which is more effective in distinguishing between self-identified gay and heterosexual male speakers of American English? Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Previous research (Tracy & Satariano, 2011) investigated how listeners were able to distinguish between self-identified gay and heterosexual male speakers of American English. In one experiment, listeners heard greater and greater portions of utterances that followed a CVC pattern (/m/, /mæ/, and /mæs/), and their judgments significantly improved upon hearing the second phone. It was unclear if this result was due to the second phone being a vowel or because it was an additional phone. It was hypothesized that judgments would improve if listeners heard additional phones. Furthermore, listeners would primarily rely on vowels, and not consonants, to form their judgments. This hypothesis was tested in the present experiment. Listeners heard utterances that contained one, two, or three consonants, and one, two or three vowels. The results demonstrated that sexual orientation judgments improved if the utterances contained three phones compared to one phone. It was also discovered that the listeners were better able to distinguish between the gay and heterosexual speakers if the utterances contained vowels rather than consonants. Thus, the hypotheses were confirmed. While sexual orientation judgments improved if the utterances contained more phones, listeners were better able to distinguish between the speakers if the utterances contained vowels.

4pSC25. The role of social information in cognitive processing: Sex and sexuality. Eric Wilbanks (English, North Carolina State Univ., 204 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695, ewwilban@ncsu.edu)

Exemplar models of cognitive linguistic processing hold that humans' robust memory faculty allows for the construction of "exemplars" or prototypes gained through statistical procedures applied to experiential memories of variation (Pierrehumbert, 2006). Since they are based upon experiential memories, exemplar clouds are sensitive to social information. As Mendoza-Denton *et al*. (2003) note, congruence between the variant and the center of the exemplar cloud often facilitates cognitive processing. Both speaker age (Walker and Hay, 2011) and speaker sex (Sumner and King, 2013) have been shown to affect semantic processing. Expanding upon these investigations, the current study examines a new social variable, sexuality. First, a lexical association task was carried out to construct a corpus of word-pairs whose semantic links differed for straight female, male, and gay male speakers. Then, the effect of congruence between speaker sexuality, gender, and semantic pair was investigated in a lexical decision task. Statistical analyses of reaction times illustrate significant increases in processing when congruence between speaker and associated pair is negative. Additionally, the effect of (in)congruence on semantic priming was greatest for the gay male speaker. These data demonstrate the saliency of sexuality information in lexical processing and semantic linking and argue for an expansion of the social categories relevant for Exemplar models of linguistic processing.

4pSC26. Phonetic convergence during conversational interaction and speech shadowing. Jennifer Pardo, Adelya Urmanche, Sherilyn Wilman, Jaclyn Wiener, Hannah Gash, Sara Parker, Keagan Francis, and Alexa Decker (Psych., Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, pardo@optonline.net)

Phonetic convergence has been studied in both speech shadowing tasks and in conversational interaction. In both settings, phonetic convergence has been found to be highly variable, with higher convergence measures usually found in studies that used speech shadowing. In order to examine whether phonetic convergence in both settings arises from similar mechanisms, the current study compared phonetic convergence in both speech shadowing and in paired conversational interaction in the same set of talkers. A set of 96 talkers (48 female) provided shadowed recordings and participated in a paired conversational map task. Moreover, the pairs were constructed to permit comparisons of same- and mixed-sex pairings. Phonetic convergence was assessed in both tasks using AXB perceptual listening tests with naïve listeners. Overall, phonetic convergence was highly variable across pairings in both shadowing and conversational tasks, with interesting effects of talker and pair sex.

4pSC27. Phonetic convergence in multiple features. Chelsea Sanker (Linguist, Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, cas443@cornell.edu)

People's speech shifts during conversation and other interactions to become more similar to the speech of interlocutors in several linguistic characteristics. Most studies on this convergent shift analyze one or two phonetic features (e.g., Babel 2009, Nielsen 2011) or holistic perceptual similarity ratings (e.g., Goldinger 1998). This study focuses on analyzing how the amount of change in difference between speakers' averages in one feature is correlated with that pair's change in difference in other features. Convergence was compared for eight speech features over the course of partners' interactions: F1, F2, vowel duration, pitch, amplitude, turn duration, duration of pauses within turns, and duration of pauses between turns. Among the 28 correlations between convergence in different features, there were only three correlations which reached significance: between within-turn pause duration and between-turn pause duration, between pitch and F2, and between pitch and turn duration, all of which can be attributed to feature correlation independent of change. The degree to which convergence is exhibited by a pair in each feature seems to be highly influenced by the individual speakers; there was a significant correlation in the convergent change within each feature between pairs including the same individual.

4pSC28. Source versus spectral cues in the perception of indexical features in speech. Ewa Jacewicz, Robert A. Fox, and Hannah Ortega (Dept. and Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu)

Spoken language includes two different forms of information: linguistic (message related) and indexical (related to individual speaker

characteristics). This study explores the nature of the acoustic cues that listeners may use to identify gender and dialect. Spontaneous utterances were produced by 40 speakers (20 male, 20 female) from two different regional dialects spoken in central Ohio (OH) and western North Carolina (NC). These utterances were equally divided into three sets of tokens. One set was unprocessed (except for amplitude equalization). A second set was low-pass filtered at 400 Hz, retaining prosodic information, but little content. A third set was processed through an 8-channel noise vocoder eliminating all harmonic information (simulating cochlear implant processing). These three stimulus sets (blocked and randomized) were played to 20 OH listeners who identified whether the token was produced by a man or a woman, from OH or NC. Gender identification rates were high (means $>$ 89%) across all three conditions with clear $>$ LP filtered $>$ vocoded. The rates for dialect identification were significantly lower overall with the LP-filtered condition close to chance (58%). Discussion will center on listener use of acoustic features and perceptual sensitivity (d') to gender and dialect.

4pSC29. Talker variation and systematicity in voice onset time: A corpus study. Eleanor Chodroff and Colin Wilson (Cognit. Sci., Johns Hopkins Univ., Krieger Hall 237, 3400 N. Charles St., Baltimore, MD 21218, chodroff@cogsci.jhu.edu)

Previous research has demonstrated variation across talkers in the phonetic realization of speech sounds, including vowels (e.g., Peterson and Barney, 1952), fricatives (e.g., Newman *et al*., 2001) and stop consonants (e.g., Allen *et al*., 2003; Theodore *et al*., 2009). The challenge that talker variation presents to perceptual processes may be significantly reduced if different aspects of talker-specific phonetics are strongly correlated. The present study employs a subset of read speech from the Mixer-6 corpus (Brandschain *et al*., 2013), containing recordings from 129 native English participants (60 male). The voice onset time (VOT) of prevocalic word-initial stops (N = 59,075) was measured using forced alignment (Yuan & Liberman, 2008) and subphonemic postprocessing (AutoVOT; Sonderegger & Keshet, 2012). Talker means of the voiceless stops are highly correlated with each other (rs = /p/-/t/: 0.76; /p/-/k/: 0.79; /t/-/k/: 0.74; $ps < 0.0006$, alpha-corrected), as are /b/ and /g/ among the voiced stops ($r = 0.49$, $p < 0.0006$). Additionally, talker means of each stop category are correlated with corresponding standard deviations ($r = 0.87$, $p < 0.0001$). These suggest a degree of systematicity in variation, with implications for talker adaptation, as listeners may extrapolate from few input values to approximate talker-specific variance and cross-category means.

4pSC30. Examining correlations between phonetic parameters: Implications for forensic speaker comparison. Erica Gold (Linguist and Modern Lang., Univ. of Huddersfield, Heslington, York YO10 5DD, United Kingdom, erica.gold@york.ac.uk) and Vincent Hughes (Lang. and Linguistic Sci., The Univ. of York, York, United Kingdom)

The research presented in this paper builds upon a previous pilot study (Gold and Hughes 2012). This paper explores the correlation structure of speech parameters from a sociolinguistically homogeneous set of male speakers of Southern Standard British English using a series of segmental, suprasegmental and linguistic parameters. Data was extracted from a subset of speakers from the Dynamic Variability in Speech (DyViS) database (Nolan *et al*., 2009) and consist of: midpoint F1, F2 & F3 values for /a ɔ u/, midpoint F1, F2 & F3 values hesitation markers UM and UH, dynamic F1, F2 & F3 values for PRICE /aɪ/, long-term formant distributions (LTFD) F1-F4, mean and standard deviation of fundamental frequency (F0), mean articulation rate (AR), voice onset time (VOT) for word-initial /t/ and /k/, and click rate (ingressive velaric stops). The results of the study present a complex correlation structure between linguistic-phonetic variables, and not all correlations are predicted by phonetic theory. The results of the correlations are discussed in relation to implications that exist when combining parameters for forensic speaker comparison casework; specifically, the caution that needs to be yielded by experts in casework to avoid over- or under-estimating the strength of evidence.

4pSC31. The relationship between fundamental frequency and within-speaker vowel reduction. Christina Kuo (Commun. Sci. and Disord., James Madison Univ., MSC4304, 801 Carrier Dr., Harrisonburg, VA 22807, kuocx@jmu.edu), Gary Weismer (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Madison, WI), and Casey Behre (Commun. Sci. and Disord., James Madison Univ., Harrisonburg, VA)

The contribution(s) of fundamental frequency (F0) to vowel production has potential theoretical and clinical implications. The sufficient contrast hypothesis [Diehl *et al*., J. Phon. **24**, 187–208 (1996)] suggests that a high F0 is associated with effects of spectral undersampling, and thus may require more exaggerated formant frequencies to maintain the perceptual distinctiveness of vowels. Nonetheless, studies on the interplay between F0 and vowel acoustics would appear inconclusive [Byrd, J. Acoust. Soc. Am. **92**, 593–596 (1992)] [Weirich & Simpson, J. Acoust. Soc. Am. **134**, 2965–2974 (2013)]. This study evaluated the sufficient contrast hypothesis as a within-speaker mechanism, with an overarching hypothesis that formant frequency exaggeration associated with spectral undersampling, if any, occurs within a given production system (i.e., speaker). To this end, formant frequencies of vowels and F0 were obtained for the same speaker across different speaking tasks. Within speaker, it was hypothesized that vowel formant frequencies as well as F0 change across tasks. More importantly, average F0 is hypothesized to be negatively correlated with the degree of vowel formant frequency reduction within-speaker across tasks. That is, a vowel system produced with a higher F0 would be less prone to reduction because the spectral peaks are better defined.

4pSC32. Within- and between-talker variability in voice quality in normal speaking situations. Jody Kreiman (Head and Neck Surgery, UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, jkreiman@ucla.edu), Patricia A. Keating (Dept. of Linguist, UCLA, Los Angeles, CA), Soo Jin Park (Dept. of Elec. Eng., UCLA, Los Angeles, CA), Shaghayegh Rastifar (Head and Neck Surgery, UCLA, Los Angeles, CA), and Abeer Alwan (Dept. of Elec. Eng., UCLA, Los Angeles, CA)

Increasing evidence suggests that voices are best thought of as complex auditory patterns, and that listeners perceive and remember voices with reference to a “prototype” or “average” for that talker. Little is known about how, and how much, individual talkers vary their voice quality across situations that arise in every-day speaking, so the nature and extent of variability underlying these abstract averages, and thus the nature of the averages themselves, is unclear. The theoretical relationship between acoustic similarity and confusability in the context of a prototype model also remains unclear. In this preliminary study, 9 tokens of the vowel /a/ were recorded from 5 females on three dates. Measures of F0, spectral slope, HNR, and formant frequencies and their variability were gathered for all voice samples and acoustic distances between talkers were calculated under the assumption that all acoustic variables were equally important perceptually. Perceptual confusability was assessed in a same/different task, and predictions under the equal perceptual weight assumption were tested. Discussion will focus on how much variability is required before a voice sample no longer sounds like the originating talker, and on how the perceptual importance of each acoustical variable varies across talkers and acoustic contexts. [Work supported by NSF and NIH.]

Session 4pUW

Underwater Acoustics: Three-Dimensional Underwater Acoustics Models and Experiments II

Ying-Tsong Lin, Cochair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543

Marcia J. Isakson, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78713

Invited Papers

1:25

4pUW1. Out-of-plane effects in three-dimensional oceans. Michael B. Porter (HLS Res., 3366 N. Torrey Pines Court, Ste. 310, La Jolla, CA 92037, mikeporter@hlsresearch.com)

The two-dimensional beam tracing algorithms in the BELLHOP acoustic model have been extended to three dimensions in BELLHOP3D. The new model includes virtually all of the capabilities of the 2D model, including 1) spatially varying bottom-types, 2) eigenray calculations, 3) TL calculations, and 4) time series calculations. In addition, the 3D model includes both the geometric and Gaussian beam tracing options. Finally, a simple option change allows it to switch between 2D and 3D calculations to assess the effects of horizontal refraction. This talk will discuss the algorithm, demonstrate its capabilities, and present results that illustrate when horizontal refraction is important.

1:45

4pUW2. Numerically exact three-dimensional propagation. Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com)

Solving propagation in a three-dimensional environment is one of the most challenging problems in computational physics. However, for cases where the environment possesses some kind of symmetry be it rotational or translational, the propagation problem can be simplified considerably. It can be shown that in problems where the environment is translationally invariant, the 3D wave equation can be Fourier transformed along the direction of translational symmetry to reduce it to a 2D equation for each spectral component. The 3D solution can be obtained by solving the 2D wave equation for each spectral component and performing the inverse Fourier transform. In this paper, we use the above technique to compute propagation in an ideal and a penetrable wedge. For the ideal wedge, the pressure-release boundary condition is applied to both boundaries and for the penetrable wedge, the pressure-release boundary condition is applied to the horizontal surface and the continuity of pressure and normal velocity is imposed on the sloped interface. To obtain a numerically exact solution, we use the virtual source technique to solve the 2D problem for each spectral component.

2:05

4pUW3. Coupled mode analysis of three-dimensional propagation over a cosine shaped hill. Megan S. Ballard (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

Three-dimensional propagation over an infinitely long cosine shaped hill is examined using an approximate normal mode/parabolic equation hybrid model that includes mode coupling in the out-going direction. The slope of the hill is relatively shallow, but it is significant enough to produce both mode-coupling and horizontal refraction effects. The modeling approach is described and the solution is compared to results obtained with a finite element method to evaluate the accuracy of the solution in light of assumptions made in formulating the model. Then, the calculated transmission loss is interpreted in terms of a modal decomposition of the field, and the solution from the hybrid model is compared to adiabatic and Nx2D solutions to assess the relative importance of horizontal refraction and mode-coupling effects. An analysis using a horizontal ray trace is presented to explain differences in the modal interference pattern observed between the 3D and Nx2D solutions. The detailed discussion provides a thorough explanation of the observed 3D propagation effects and demonstrates the usefulness of the approximate normal mode/parabolic equation hybrid model as a tool to understand measured transmission loss in complex environments. [Work supported by ONR.]

Contributed Papers

2:25

4pUW4. Modeling three dimensional scattering from rough ocean boundaries using finite elements. Marcia J. Isakson and Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Scattering from three-dimensional rough interfaces can be estimated using approximations to the Helmholtz-Kirchhoff integral such as perturbation theory or the Kirchhoff approximation. However, these solutions are constrained to the boundary. Physical processes occurring beneath the boundary such as scattering from layers or volume inclusions require additional approximations. Finite element models can benchmark these approximations since scattering the entire volume is calculated. In this study, a three-dimensional finite element model of acoustic scattering from rough interfaces is developed and compared with boundary element solutions. This model is an improvement on the existing two-dimensional and longitudinally invariant models previously presented. The full three-dimensional nature of this solution allows realistic representations of layering and volume inclusions, including shells, air bubbles, and targets. [Work supported by ONR, Ocean Acoustics.]

2:40

4pUW5. Three-dimensional underwater acoustic scattering from ocean boundaries: Proposed benchmark problems. Marcia J. Isakson, Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu), and Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Acoustic scattering from ocean boundaries is a major component of both the signal for mapping sonars and the noise for imaging sonars. However, models of scattering are often reliant on boundary methods such as the evaluation of the Helmholtz-Kirchhoff integral and its approximations. These methods do not account for sub-surface scattering from layers or inclusions although additional approximations are often used. Often, these approximations are employed with only a cursory knowledge of their realm of validity with respect to such parameters as interface roughness conditions, sediment types and layering structure. We propose a series of benchmark problems to begin to establish the regions of validity for extant scattering models. The problems include 3D scattering from an air/water interface, scattering from a layered ocean sediment and scattering from a simple target buried beneath a rough ocean floor. By comparing the results from a variety of benchmark models, it may be possible to establish a better range of validity for the current suite of modeling tools. [Work supported by ONR, Ocean Acoustics.]

2:55

4pUW6. Three-dimensional underwater sound propagation: Proposed benchmark problems. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu) and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

A set of benchmark problems is suggested for testing the accuracy and efficiency of numerical solutions for three-dimensional (3D) underwater sound propagation. The physical causes of the 3D propagation conditions considered in this talk include out-of-plane reflections from medium boundaries and interfaces and horizontal refractions caused by changes of the index of refraction in the medium. Five different propagation scenarios are suggested: (1) sound propagation in a slope environment, (2) nonlinear internal wave waveguides, (3) submarine canyons, (4) seamounts, and (5) 3D waveguides with surface waves and/or seafloor sand waves. Both idealized and realistic environments will be considered. To allow theoretical closed-form solutions, the model environments will be simplified to create idealized problems, such as an impenetrable wedge/slope model and a semi-circular waveguide with impenetrable boundaries. For the problems which do not have closed-form solutions, different numerical solutions will be generated for comparisons. The numerical methods focused in this talk include 3D parabolic-equation, normal mode and finite element methods, and discussions on other numerical techniques, such as ray tracing, wavenumber integration, finite difference, and boundary element methods, will also be provided. [Work supported by the ONR.]

3:10–3:25 Panel Discussion

3:25–3:40 Break

Invited Paper

3:40

4pUW7. Links between acoustic field statistics, baroclinic wave geometry, and bathymetry computed with time-stepped three-dimensional acoustic simulations. Timothy F. Duda, Arthur E. Newhall, and Ying-Tsong Lin (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu)

Simulations of sound propagation in a canyon region and in an area populated with internal-wave packets have been performed. Time dependence was introduced into the three-dimensional water-column environments using dynamical ocean models. In the canyon simulation, time-dependence is limited to waves with hydrostatic pressure generated by tidal transports prescribed at the dynamical model boundaries. The wave-packet simulation includes both hydrostatic and nonhydrostatic waves. For the canyon, propagation of 300-Hz sound from shallow to deep water is simulated. For wave packets, 200-Hz sound is simulated. For each situation, statistics related to horizontal correlation are estimated at synthetic "arrays" inserted into the domain. Correlation properties are estimated from single field snapshots and from time-series analysis of the output, and the two are compared. Array performance evaluations from the two methods are also compared. Covariance matrix properties for the time-series output, including principal component behavior, are analyzed to examine the interference effects and fluctuation mechanisms that are responsible for the spatially patchy array performance results.

4:00

4pUW8. Three-dimensional parabolic-equation solutions with boundary-fitted grids. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

A higher-order operator splitting method incorporating multi-dimensional cross terms has recently been proposed to solve the three-dimensional (3D) parabolic wave equation consisting of a square-root Helmholtz operator. The advantages of this splitting method include providing a more accurate 3D parabolic-equation (PE) approximation, as well as supporting fast marching solvers, such as the Alternating Direction Implicit (ADI) Padé method. To apply this numerical solution scheme to a 3D underwater acoustic waveguide with surface waves in a boundary-fitted model grid, one can employ a one-dimensional (1D) non-uniform discretization formula derived from Galerkin's method using asymmetric basis functions [W.M. Sanders and M.D. Collins, J. Acoust. Soc. Am. **133**, 1953–1958 (2013)]. The use of this discretization formula is to approximate the two alternating sets of 1D differential equations with respect to either one of the two transverse directions at the ADI marching steps. An idealized semi-circular waveguide problem with a closed-form solution is considered as a benchmark to test different 3D PE solutions. Compared to the fixed grid PE solution, the boundary-fitted PE solution is an order of magnitude more accurate in terms of both the magnitude error per unit distance and the relative phase error. [Work supported by the ONR.]

4:15

4pUW9. Three-dimensional noise modeling in a submarine canyon. David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., PO Box 15000, Halifax, Nova Scotia B3H 4R2, Canada, dbarclay@dal.ca) and Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The ambient sound field due to wind generate surface noise in an idealized Gaussian submarine canyon can be described using the method of normal mode decomposition applied to a three-dimensional longitudinally invariant wave-guide. The modal decomposition is carried out in the vertical and across-canyon horizontal directions and gives a semi-analytical solution describing the three-dimensional topographic effects on the spatial distribution of the noise, the power spectrum, and the vertical and horizontal coherences. Additionally, the noise field can be computed using wave-equation reciprocity and either a three-dimensional cylindrical co-ordinates parabolic equation (PE) or an Nx2D PE sound propagation model. Inter-comparison of these models highlights the effect of the three-dimensional topography on the vertical coherence and mean-noise level as a function of arrival direction relative to the canyon's axis. These effects include the focusing of noise along the canyon axis and the frequency perturbation of vertical coherence minima.

4:30

4pUW10. Time-domain broadband simulations of arbitrarily shaped three-dimensional elastic targets on water/sand interface. Yu Shao, Myoung An, and Shumin Wang (Auburn Univ., 200 Broun Hall, Auburn, AL 36849, wangs@auburn.edu)

Acoustic wave scattering from buried or proud elastic targets on ocean floor is important to a number of engineering problems. Existing simulation approaches are mainly based on frequency-domain finite-element method (FEM). Although accurate, it is inefficient for broadband acoustic signals for two main reasons. First, the computational cost is proportional to the number of frequency points of interests. Second, inversion of the resulting FEM matrix at each frequency point can be quite time-consuming as the number of unknown increases. The latter can be especially problematic at higher frequencies as the spatial resolution or element order increases. A staggered-grid finite-difference time-domain (FDTD) method is developed to overcome these problems. It simulates the time-domain signals directly in a leap-frog fashion without the needs of explicitly storing and inverting system matrices. The perfectly matched layers (PML) technique is applied to truncate the simulation domain and the total-field/scattered-field approach is employed to incorporate incident, scattered, and transmitted waves from the water/sand interface. Finally, the far-field scattered field is computed via the layered Green's function from the near-field results. Several three-dimensional numerical examples are further provided to demonstrate its accuracy and efficiency.

4:45

4pUW11. Parameter dependence of normal modes for a coastal front model. Brendan J. DeCourcy (Mathematical Sci., Rensselaer Polytechnic Inst., 1521 6th Ave., Apartment 313, Troy, NY 12180, decoub@rpi.edu), Ying-Tsong Lin, James F. Lynch (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

A feature model for a shallow ocean front over a bottom with constant slope [Y.-T. Lin and J. F. Lynch, J. Acoust. Soc. Am. **131**, EL1–EL7 (2012)] is analyzed to determine the parameter dependence of three-dimensional normal mode solutions. The front is a curved interface between two isospeed regions in a coastal wedge. Relevant parameters are the front distance from the straight shoreline, bottom slope angle, sound speeds, and source frequency. The cross-slope wavenumbers are determined by a complicated dispersion relation involving Hankel functions, so a series of asymptotic approximations is applied to simplify the equation. The result is converted into a first-order differential equation for the wavenumber variation with respect to a selected parameter, and accurate and explicit solution expressions are obtained. These expressions permit convenient specification of how model parameters influence acoustic quantities such as modal phase speed. This approach is designed for application to other ocean feature models which contain large or small dimensionless combinations of parameters. The objectives are to help explain how acoustic metrics depend on environmental and geometrical parameters, and to assist in assessing results from integrated ocean-acoustics models. [Work supported by the ONR.]

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday and Thursday evenings. On Tuesday, the meetings will begin at 7:30 p.m., except for Engineering Acoustics, which will hold its meeting starting at 4:30 p.m. On Thursday evening, the meetings will begin at 8:00 p.m. or 8:30 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 Thursday are as follows:

8:00 p.m.	
Biomedical Acoustics	Kings 2
Musical Acoustics	Brigade
Noise	King 5
Speech Communication	Commonwealth 2
Underwater Acoustics	Ballroom 4
8:30 p.m.	
Signal Processing in Acoustics	Ballroom 3