Session 3aAAa

Architectural Acoustics: J. Christopher Jaffe—His Life in Acoustics

K. Anthony Hoover, Cochair
McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362

William J. Cavanaugh, Cochair
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Alexander U. Case, Cochair
Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854

Chair's Introduction—8:25

Invited Papers

8:30

3aAAa1. Recollections on Chris Jaffe: From his early days in ASA through an enduring friendship of nearly six decades. William J. Cavanaugh (Cavanaugh Tocci Assoc. Inc., 327F Boston Post Rd., Sudbury, MA 01776, wcavanaugh@cavtocci.com)

In some 60 years of consulting in architectural and environmental acoustics, one encounters few colleagues who fit the definition of a truly renaissance man. J. Christopher Jaffe is one of those few. His love for the performing arts and in designing facilities for the enjoyment of countless audiences and performers alike knew no bounds. My friendship with Chris began with hearing the very first paper he presented at an ASA meeting on the frequency selective properties of stage enclosures and extended throughout his entire professional career. Chris Jaffe epitomizes the challenge stated in the by-laws of our Society: to spread knowledge in acoustics and to promote its practical applications. Chris lived this challenge for a too short professional life with unabated enthusiasm and joy. We miss him terribly but are comforted that he made the world a better place.

8:45

3aAAa2. J. Christopher Jaffe: Scientist and friend. Leo L. Beranek (Retired, 10 Longwood Dr., Westwood, MA 02090, beranekleo@ieee.org)

Chris Jaffe is acclaimed for his original designs of spaces for the performance of music. It has been this author/s pleasure to discuss his many projects and to hear concerts in his four best known halls: Severance Hall in Cleveland, Ohio; Bass Performance Hall in Fort Worth, Texas; the Concert Hall at the Kennedy Center in Washington, D.C.; and the Sala Nezahualcoyotln in Mexico City. Chris took great pleasure in teaching others the field and was the leader in establishing the Concert Hall Research Group that took data in many halls and held seminars for acousticians at all levels. Other facets in his life will be discussed.

9:00

3aAAa3. Remembrances of Chris Jaffe—An innovator in many fields, our colleague, and friend. Wade Bray (HEAD Acoust., Inc., 6964 Kensington Rd., Brighton, MI 48116, wbray@headacoustics.com), Mahlon D. Burkhard (Retired, Adamstown, MD), and Klaus Genuit (HEAD Acoust. GmbH, Herzogenrath, Nordrhein-Westfalia, Germany)

A remembrance of Christopher Jaffe in the technical perspective will be given, in a matrix of the human perspective of working with him, learning from him and enjoying his vision, friendship, and wit. The authors all collaborated with Chris: Wade Bray as a consultant at Jaffe Acoustics, Mahlon Burkhard as an acoustical colleague and developer of electronic technologies proposed by Chris through the major era of the Electronic Reflected Energy System (ERES) starting with the "Live from Studio 8H" NBC television symphonic concerts—birth of the Chris-named "NBC Delay"—and Klaus Genuit fulfilling Chris's vision of binaural technology in multiple uses—Chris providing the avenue for HEAD acoustics GmbH to begin serving North America. Christopher Jaffe was a thorough, novel acoustician: fluent and innovative in both non-electronic and electronic techniques, though perhaps better known for his lifelong passion about the latter in the pure service of the listening experience and its reaching more people. He had a unique personal, charismatic presence, an ability not only to explain technical acoustics in artistic context to nontechnical people, but also to enfold them in the enthusiasm, skill and momentum of his spirit. We celebrate him; we miss him, his path lives.

9:15

3aAAa4. Chris Jaffe's contributions to the Concert Hall Research Group. Timothy Foulkes (Cavanaugh Tocci Assoc., 327 Boston Post Rd., Sudbury, MA 01776, tfoulkes@cavtocci.com)

One of Chris Jaffe's many contributions to acoustics was his involvement in the Concert Hall Research Group (CHRG). In 1992, Chris had the idea of sending three measurement teams to the same concert halls for the dual purpose of getting current data and also to see how close the results would be between different teams measuring the same hall. From the inception of the CHRG in 1992 until a few months before his death in 2012, Chris was a source of ideas, inspiration, and energy.

9:30

3aAAa5. The Jaffe effect. Robin S. Glosemeyer Petrone (Threshold Acoust., 53 W Jackson Blvd., Ste. 815, Chicago, IL 60604, robin@thresholdacoustics.com)

By 1940, nearly every sizable city in America had a movie palace or movie-adapted vaudeville theater. As these regal houses for the silver screen were replaced by cinemas, the palaces and vaudeville theaters of the early 20th century sat with empty stages, taunting orchestras, theater companies, and would be audiences. Christopher Jaffe's most noted contribution to architectural acoustics was championing the adaptive re-use of these theaters through the development of an orchestra shell system. These light-weight shell systems allowed the theaters to support multiple performing arts organizations without a monumental capital campaign that neither the cities nor art organizations could afford. Chris' most singular contribution to acoustics was, however, his mentorship. This maverick in the field, always on a quest to prove himself, never missed an opportunity to share his knowledge or work out a new concept with a colleague or apprentice. He fostered the growth of consultants at all stages of their careers, first in his practice and later in the establishment of an architectural acoustics program at his alma mater.

9:45

3aAAa6. Christopher Jaffe and the Graduate Program in Architectural Acoustics at Rensselaer Polytechnic Institute. Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, xiangn@rpi.edu)

Chris Jaffe graduated with a major in chemical engineering in 1949 from Rensselaer Polytechnic Institute. Four decades later in 1998, Chris founded an Architectural Acoustics program at the School of Architecture, Rensselaer Polytechnic Institute. In the fields of architectural-, physical-, and psycho-acoustics, the rapid pace of change has advanced the program to be a graduate program with an ambitious mission of educating future experts and leaders in architectural acoustics. Chris Jaffe's continued dedication and support helped Rensselaer's Graduate Program in Architectural Acoustics reshape its pedagogy using "STEM" (science, technology, engineering, and mathematics) methods, including intensive, integrative hands-on experimental components that fuse theory and practice in a collaborative environment. The STEM-based pedagogy enables individuals from a broad range of fields to succeed in this rapidly changing field. The program has attracted graduate students from a variety of disciplines including individuals with B.Arch., B.S., or B.A. degrees in Architecture, Music, Engineering, Audio/Recording Engineering, Physics, Mathematics, Computer Science, Acoustics, Electronic Media, and related fields. RPI's Graduate Program in Architectural Acoustics has since graduated more than 100 graduates with both M.S. and Ph.D. degrees. This paper shares the growth and evolution of the graduate program, and acknowledges the profound contributions made by Chris Jaffe.

10:00–10:15 Break

10:15

3aAAa7. Jaffe's signature: Science, with art. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

I am one of a handful of students claiming the title, "Chris Jaffe's first graduate student,"—but I really am the first! I found at Rensselaer Polytechnic Institute a program that reflects his priorities. We did not just study the physics of room acoustics and make measurements in built spaces. We worked side by side with architects, engineers, electronic musicians, and other artists. Acoustics was not an isolated achievement. It was always part of a collaborative effort with other ambitious disciplines. We built things—big, awe-inspiring things—visually stunning, sonically thrilling. I found myself studying, measuring, and recording in all kinds of spaces, from concert halls in Texas to subway stations in New York. Thanks to Dr. Jaffe's leadership, were always doing, growing, learning, and laughing.

10:30

3aAAa8. Fortuitous coupling: A recollection of how Chris Jaffe brought together coupled rooms and Texas oil money to launch a graduate program and a career. Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

Early in his career Chris Jaffe discovered that reverberation-starved Vaudeville theaters could be improved through use of light-weight demountable shells for the stage enclosures. These shells could be designed for frequency-dependent "selective transmission" of energy into the otherwise unused stage house, realizing a "fortuitous coupling" that improved acoustics for the audience and on-stage musicians. My experience with Chris at Rensselaer Polytechnic Institute (RPI) is also one of fortuitous coupling, as he orchestrated the launch a new graduate program in architectural acoustics. Chris had just overseen the completion of Bass Performance Hall in Fort Worth, Texas, which realized the culmination of his design philosophy for stage-house coupling in multipurpose theaters. Now, Bass Hall was to become a laboratory for studying the science underlying Chris's empirical findings. RPI, Chris's alma mater, was to be the institutional home for that research. And Ed Bass, scion of the Bass family and primary benefactor of the hall's construction, would provide the initial support. As a student, about to complete my Master's degree and ready leave physics for audio engineering, a chance elective course followed by a Medici-esque offer to study the physics of coupled rooms radically shifted the course of my professional life.

10:45

3aAAa9. Chris Jaffe: Youthful golden years. Benjamin Markham and Carl Rosenberg (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

From 2009 until his demise—roughly 50 years after his entrée into the acoustics field—Chris Jaffe was a valued partner in the Acentech Studio A team. We will share the manner in which his mentorship and guidance informed us and our profession. His self-deprecating humor could not disguise his intense knowledge, broad experience, willingness to help mentor junior staff, and engagement in new endeavors. He imbued his acoustics consulting "golden years" with youthful vigor, and he accomplished much: he finished his book, consulted on projects, shared back stories, enlivened our parties, introduced us to his friends and his manner of collaborating with them, and made an indelible imprint on both the content and style of our performing arts practice. Our presentation will share these experiences as life lessons for success and respect in our profession.

11:00

3aAAa10. Chris Jaffe's three A's. Malcolm Holzman (Holzman Moss Bottino Architecture, 214 West 29th St. Tower 17fl, New York, NY 10001, mholzman@holzmanmoss.com)

The first A is for Academia: The importance Chris Jaffe put on education as a foundation and a concentration for his career. The second A is Acoustics: A lifespan focused on propelling the science and art of architectural acoustics in the civic and academic presentations of music, theater, and dance. The third A is Adirondacks: His affection and attachment to the Adirondacks were a measure of the man.

11:15–11:45 Open mic Discussion

WEDNESDAY MORNING, 7 MAY 2014 550 A/B, 9:00 A.M. TO 12:00 NOON

Session 3aAAb

Listening to the "Virtual Paul's Cross"—Auralizing 17th Century London I

Matthew Azevedo, Chair
Acentech Inc., 33 Moulton St., Cambridge, MA 02138

The purpose of this session is to provide an opportunity for people to listen to the Virtual Paul's Cross auralization, which allows listeners to experience John Donne's 1622 Gunpowder Day sermon while surrounded in three dimensions by a reactive crowd of up to five thousand, the bells of St. Paul's, and the ambient soundscape of 17th century London. The auralization allows for real-time changes to crowd size, listener position, the behavior of the intelligent agents which create the crowd reactions, and variations in the type and frequency of ambient sounds and requires over one hundred concurrent audio channels, a dozen channels of real-time convolution, hours of project-specific source recordings, and a complex network of intelligent and stochastic logical structures.

WEDNESDAY MORNING, 7 MAY 2014 554 A/B, 9:00 A.M. TO 12:00 NOON

Session 3aAB

Animal Bioacoustics: Sound Production and Reception by Animals

James A. Simmons, Chair
Neuroscience, Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912

Contributed Papers

9:00

3aAB1. Behavioral analysis of lateral line and vestibular hair cell function in developing *Xenopus laevis*. Andrew T. Stevens-Smith and Andrea Simmons (Brown Univ., Providence, RI 02912, Andrew_T_Smith@brown.edu)

Xenopus tadpoles are aquatic, nocturnal animals with well-developed superficial lateral line neuromasts. We previously showed that these animals, from premetamorphic through early froglet stages, exhibit stereotyped responses to low velocity flow fields—they move downstream at flow onset, and then turn and orient towards the flow source (positive rheotaxis). They continue to station hold in this oriented position for the duration of the stimulus. Positive rheotaxis and station holding are disrupted, but not totally eliminated, when animals are exposed to high concentrations of the ototoxic drug gentamicin. These data suggest that another sensory system is involved in detection of current flow. To determine the role of vestibular hair cells in these behaviors, we injected gentamicin directly into the tadpoles' otic capsules. This procedure damaged the developing otoliths and resulted in abnormal circular swimming movements and increased latency of rheotaxis and station holding. These behaviors were not, however, completely disrupted. These data suggest that the lateral line and vestibular systems act together in mediating responses of Xenopus tadpoles to flow fields.

9:15

3aAB2. Selection on aerial hearing in turtles: Auditory evoked potentials in the box turtle, *Terrapene carolina* relative to the stinkpot, *Sternotherus odoratus*. Jeffrey Zeyl and Carol E. Johnston (Fisheries, Aquaculture and Aquatic Sci., Auburn Univ., 203 Swingle Hall, Auburn University, Auburn, AL 36849, jnz0002@tigermail.auburn.edu)

Mechanisms of auditory stimulation differ underwater versus in air due to differences in each medium's acoustic impedance, which has resulted in unique hearing adaptations in aquatic relative to terrestrial organisms. Testudines are a useful taxon for studying the evolution of hearing specializations in relation to the air-water interface because this group includes members at various points on the aquatic-terrestrial lifestyle continuum. Here we tested for differences in auditory function between the terrestrial box turtle, *Terrapene carolina* (Emydidae) and the fully aquatic stinkpot, *Sternotherus odoratus* (Kinosternidae). Auditory evoked potentials were collected in response to tone pips to generate threshold audiograms in air as well as with tympana submerged underwater. Sensitivities and bandwidths of both species were similar underwater and in air, but the thresholds of box turtles were 9–20 dB more sensitive than stinkpots in air across the entire frequency range. The results indicate selective pressures to enhance aerial hearing in box turtles.

9:30

3aAB3. Underwater signal attenuation of northern red-legged frog calls. Jodi Gronborg (Biology, Portland State Univ., 1719 SW 10th Ave., SRTC 246, Portland, OR 97201, gronborg@pdx.edu)

Shallow water acoustics can dramatically alter spectral profiles of the northern red-legged frog underwater advertisement calls and should be taken into consideration when designing man-made aquatic environments as part of habitat mitigation. While much is known about atmospheric constraints on acoustic communication, we need to learn more about aquatic constraints. What is known about underwater acoustic signal transmission is garnered primarily from deep ocean research, leaving much to be discovered about the relatively extreme shallow underwater environments such as exists in ponds. The northern red-legged frog, *Rana aurora*, is one of three ranid species shown to vocalize while submerged underwater, and only one of two known to use the underwater portion of its environment for mating. My hydrophone recordings of the underwater chorus include calls with dominant frequencies in the 5000—15000 Hz range, sharply contrasting with the established dominant frequency range for advertisement calls of the northern red-legged frog of 450–1300 Hz. I conducted frequency sweep playbacks at breeding which demonstrate that shallow residential water bodies have unique frequency responses that dramatically alter the spectral profiles of underwater signals. This changes the frequency characteristics of what is heard by females and could significantly impact reproductive success.

9:45

3aAB4. Stimulus-frequency and response timing in clouded leopards: Evidence for inner ear adaptation. Edward J. Walsh (Developmental Auditory Physiol. Lab, Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, edward.walsh@boystown.org), Christopher A. Shera (Eaton-Peabody Labs., Harvard Med. School Massachusetts Eye & Ear Infirmary, Boston, MA), Carolina Abdala (Dept. of Otolaryngol., Univ. of Southern California, Los Angeles, CA), Heather E. Robertson (Nashville Zoo at Grassmere, Nashville, TN), and JoAnn McGee (Developmental Auditory Physiol. Lab, Boys Town National Res. Hospital, Omaha, NE)

Unlike the inverse relationship between response timing (latency) and stimulus frequency commonly observed in other fields, the relationship in species belonging to the Pantherinae subfamily is more complex. Although an inverse relationship exists between neural timing and low frequency stimuli (between 0.5 and 1–2 kHz), the relationship for frequencies greater than 2 kHz breaks from this pattern and is best described by a positive-going parabola-like curve with a maximum value in the vicinity of 8 to 16 kHz in representatives of the Pantherinae lineage studied thus far. In clouded leopards the local maximum is near 8 kHz. This general latency-frequency pattern has been confirmed in tigers and appears to hold for jaguars and lions. Clouded leopards (*Neofilis nebulosa*) branched directly from the Panthera lineage approximately 6 million years ago. This phylogenetic proximity to Panthera makes the genus Neofilis particularly interesting in relation to the response timing question. The outcome of previous efforts to determine the cochlear site of origin of short latency responses to lower frequency stimuli in the tiger were consistent with the existence of a basal turn timing adaptation. The implications of such adaptation in clouded leopards will be considered.

10:00

3aAB5. A subtraction technique for removing playback noise from high-frequency rodent recordings. Mustafa Z. Abbasi (Appl. Res. Lab. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78751, mustafa_abbasi@utexas.edu), Bret Pasch, Amilia Humber, Michael J. Ryan (Dept. of Integrative Biology, Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Appl. Res. Lab. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

The efficacy of animal communication often necessitates signalers to adjust signals. For example, many vertebrates modify vocal output to minimize interference from ambient noise (e.g., Lombard effect) or in response to other signaling animals. Such auditory-feedback-mediated vocal control is well documented in songbirds, humans, and nonhuman primates, but has not been explored in rodents. Alston's singing mouse (*Scotinomys teguina*) emit advertisement vocalizations that function in mate attraction and male-male aggression, and can vary both temporal and spectral features depending on the social context. In this experiment, we investigated the extent to which mice can modify vocal output in response to perturbations in auditory feedback by broadcasting conspecific vocalizations that overlapped a focal male's song. However, an unexpected challenge was found in separating the mouse's vocalization from the broadcast stimuli. Other studies have used Golay codes to measure the impulse response of the system and subtract the noise. However, such techniques could not be applied herein due to the high frequencies produced by mice; movement of the mouses' head appears to change the system sufficiently to prevent effective subtraction. The authors will present a novel method using spectral envelopes and cross-correlation procedures to garner feedback on the validity of this technique.

10:15–10:30 Break

10:30

3aAB6. Frequency and intensity difference limens in mice. Katrina Toal, Kelly E. Radziwon, and Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, 231 Affinity Ln., Apt. F, Buffalo, NY 14215, ktoal@buffalo.edu),

The ability to distinguish between different frequencies and intensities is a fundamental auditory process, yet it has not been explored in CBA/CaJ mice. Previous researchers have used behavioral approaches to examine frequency and intensity discrimination abilities of the NMRI mouse, feral house mice, and other strains of mice with known hearing impairments. The

present experiments use operant conditioning procedures to investigate the basic hearing abilities of the normal-hearing CBA/CaJ mouse. Intensity difference limens (IDLs) were obtained for six subjects and frequency difference limens (FDLs) were obtained for three subjects. For both experiments, the Method of Constant Stimuli and a threshold d' of 1.5 were used. IDLs and FDLs were obtained for 12, 16, 24, and 42 kHz tones. FDLs were obtained at 10 dB SL and 30 dB SL whereas the background for the IDL task was only 10 dB SL. At higher frequencies, the calculated FDLs increased. Furthermore, the thresholds were higher when sounds were presented at 30 dB SL compared to 10 dB SL. In the FDL experiment, the mice had a mean just-noticeable-difference of 3.5% Weber fraction across all four frequencies and sound levels. Interestingly, IDLs were similar across all frequencies.

10:45

3aAB7. Twists and turns, in cochlear anatomy: Curvatures related to infra vs ultrasonic hearing. Darlene R. Ketten (CMST, Appl. Phys. and Imaging, Curtin Univ./Harvard Med. School, CMST, GPO Box 1987, Perth, WA 6845, Australia, dketten@whoi.edu), James Simmons (NeuroSci., Brown Univ., Providence, RI), Hiroshi Riquimaroux (Graduate School of Life and Medical Sci., Doshisha Univ., Kyoto, Japan), Scott Cramer and Julie Arruda (Biology, Woods Hole Oceanogr. Inst. , Woods Hole, MA)

Microchiropteran bats and odontocete cetaceans are sophisticated echolocators with acute ultrasonic hearing operating in radically different media. Similarly, elephants and mysticetes share the ability to generate and respond to infrasonics. In this study, the heads, outer, middle, and inner ears of 32 specimens from 11 species of bats, dolphins, elephants, and whales were analyzed with microCT (11 to 100 micron isotropic voxel imaging; Siemens Volume Zoom and X-Tek CT units). Canal length, basilar membrane dimensions, and cochlear curvatures varied widely among all species. Length correlates with body mass, not hearing ranges. High and low frequency limits correlate with basilar membrane ratios and radii ratios, which are a measure of the radius of curvature. The ears of the known echolocators were significantly different from the mid to low frequency ears, with increased stiffness, thicker membranes, and outer osseous laminae supporting up to 60% of the basilar membrane. Anatomical correlates of "foveal" regions with stretched representation for peak echolocation spectra were found in both bat and porpoise ears. Radii and membrane ratios are consistent despite media and are predictive of high and low frequency hearing limits in all ear types. [Work supported by NIH, JIP, N45/LMRS -US Navy Environmental Division, and ONR Global.]

11:00

3aAB8. Enhance beam formation by airsacs and skull in Chinese river dolphin (*Lipotes vexillifer*). Chong Wei (College of Ocean & Earth Sci., Xiamen Univ., Hawaii Inst. of Marine Biology, Lilipuna Rd., Kaneohe, Hawaii 96744, chongwei@hawaii.com), Whitlow Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii at Manoa, Kaneohe, HI), Zhongchang Song and Yu Zhang (Key Lab. of Underwater Acoust. Commun. and Marine Information Technol. of the Ministry of Education, Xiamen Univ., Xiamen, Fujian, China)

The melon of dolphins is considered by many as the structure responsible for the focusing of the biosonar beam. However, finite element numerical simulation of the head of the Chinese river dolphin (*Lipotes vexillifer*) indicates that the biosonar beam is formed by reflections off the airsacs and bony structures in the skull. The finite element approach was applied to numerically simulate the acoustic propagation through dolphin's head in four several situations (complete head, skull only, skull plus melon, and skull plus airsacs). The acoustic intensity distribution and the corresponding polar plots showed that the melon causes the beam to narrow slightly and affects the angle of the main beam. The airsacs kept the sound propagating to the anterior and focused the energy into the main lobe. The bony structure prevented the sound from propagating below the rostrum and contribute to energy in the main beam. The results suggest that the airsacs and the complex bony structure play a dominant role in the formation of the biosonar beam of a dolphin, more so than the melon.

11:15

3aAB9. *In vivo* ultrasonic attenuation in cetacean extracranial soft tissues. Michael D. Gray, Peter H. Rogers, Peter J. Cameron (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332-0405, michael.gray@me.gatech.edu), and Gregory D. Bossart (Georgia Aquarium, Atlanta, GA)

In vivo ultrasonic attenuation was estimated for extracranial soft tissues of two *Tursiops truncatus* and one *Delphinapterus leucas*. Backscatter data were non-invasively collected as part of routine health-based ultrasound examinations using a transducer operating in the 2.0–3.5 MHz frequency range. Data sets collected over the proximal mandible and temporal regions were processed to yield estimates of attenuation using a reference tissue phantom whose properties had been independently determined. The estimated attenuations were at the low end of the range of reported values for *in vitro* mammalian fatty and connective tissues.

11:30

3aAB10. Numerical simulation of sound generation by Cicada. Derke Hughes (Sensors and SONAR, NUWCDIVNPT, 1176 Howell St., Newport, RI 02841, derke.hughes@navy.mil), Allan D. Pierce (Retired, East Sandwich, MA), Richard A. Katz (Sensor and SONAR, NUWCDIVNPT, Newport, RI), and Robert M. Koch (Chief Technol. Office, NUWC, Newport, RI)

The principal anatomical structures in the cicada that radiate sound are two platelets referred to as tymbals, which vibrate after being struck by ribs that have undergone buckling. This research effort investigates the sound of these ribbed finite plates connected to a parallel surface by a nonlinear spring. When individual ribs are placed under compression, the linearized version of the model predicts eventual exponential growth of the transverse displacement when the compressional load exceeds the buckling load. The nonlinear spring, however, stops this growth and a subsequent oscillation ensues. The actual anatomy of the cicada is more complicated than this basic model. However, this simplified mathematical explanation is given as a means to describe sound emitted in a sequence of closely spaced tone bursts. The energy from these sound impulses are stored in tensed muscles and released via buckling into the kinetic energy of ribs, which is similar to striking a drum. The tymbals "ring" at a frequency controlled by the mass of the tymbals and the air cavity "spring" within the abdomen. This ringing vibration affects the amplitude, cycles within each pulse, and the damping of the tymbal function to generate the efficiency of the cicada sound radiation.

11:45

3aAB11. A comparison of hees and haws: Donkey, Grevy's zebra, and African penguin. David Browning (139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com) and Peter Scheifele (Dept. of Comm. Sci., Univ. of Cincinnati, Cincinnati, OH)

A surprisingly few creatures bray (vocalizing during both breath in and breath out). It is not apparent why this is so or what advantage it gives those that do. This is a comparison of the technique and acoustic content for the three most well-known brayers. Donkeys (*Equus asinus*) have the most rich or raucus bray, depending on your point of view. Some donkeys start with a hee (breath inflow) and others with a haw (breath outflow), generally continuing until they are out of breath. The Grevy's zebra (*Equus grevy*) shares the perrisodactyl ability to vary frequency during vocalization but it appears to be a strained activity, resulting in a scaled down version of the donkey bray. The African penguin (*Spheniscus demersus*) has the most uniform and tonal haw, in most cases preceded by a series of short hees, apparently to increase breath.

Session 3aBAa

Biomedical Acoustics: Measurement and Imaging

Ronald A. Roy, Chair
Dept. of Eng. Sci., Univ. of Oxford, Oxford OX1 3PJ, UK

Contributed Papers

8:00

3aBAa1. Ultrasound acoustic shadow width is an accurate predictor of kidney stone size. Franklin C. Lee (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Barbrina Dunmire (Ctr. Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Jonathan D. Harper (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Bryan W. Cunitz, Marla Paun, Michael Bailey (Ctr. Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, bailey@apl.washington.edu), and Mathew D. Sorensen (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA)

Previous studies have shown overestimation of kidney stone size in ultrasound images. We explored measuring the stone's acoustic shadow as a predictor of stone size. Forty-five calcium oxalate monohydrate (COM) kidney stones ranging from 1 to 10 mm were imaged in a water bath using a research-based ultrasound system and C5-2 transducer. Stones were imaged at depths of 6, 10, and 14 cm. The widths across the stone image and across acoustic shadow distal to the stone image were measured by the operator and through an automated algorithm. Measuring size across the stone image consistently overestimated: overestimation was 0.9±0.8 mm, 1.5±1.0 mm, 2.0±1.2 mm (manual) and 0.5±1.7 mm, 0.4±1.5 mm, 0.8±1.1 mm (automated) at 6, 10, and 14 cm depths. Measurement of the acoustic shadow width more accurately estimated stone size: 0.0±0.4 mm, 0.0±0.6 mm, and-0.2±0.8 mm (manual) and 0.2±0.5 mm, 0.1±0.8 mm, and 0.1±1.0 mm (automated) at 6, 10, and 14 cm depths. Measurement from the shadow reduced misclassification of passable stones $<$5 mm to requiring surgery $>$5 mm from 25% to 7%. The results have implications for directing treatment of asymptomatic stones based on ultrasound images. [Work supported by NIH DK043881, DK092197, and NSBRI through NASA NCC 9-58.]

8:15

3aBAa2. Biofilm mitigation by ultrasound-assisted liposome treatment applied to a national aeronautics and space administration project. Junru Wu (Phys., Univ. of Vermont, 1 Whiteface St., South Burlington, VT 05403, jwu@uvm.edu), Dong Ma (School of Biological Sci. and Medical Eng., Southeast Univ., Burlington, Vermont), Graham Willsey, and Matthew Wargo (Microbiology and Molecular Genetictons, Univ. of Vermont, Burlington, VT)

Space exploration requires an effective means of biofilm control for water reclamation systems that minimizes use of non-recyclable chemicals (such as iodine and antibiotics) and does not contaminate water in the water treatment system. Uncontrolled biofilm growth due to ineffective mitigation has been the cause of water reclamation system failures both on the Russian Mir Space Station and on the International Space Station, as well as corrosion problems in thermal systems. The challenge of biofilm migration mainly comes from: its matrix of extracellular polymeric substances, in which biofilm organisms embed themselves, inhabiting bacterium in biofilm treated by chemical and mechanical stresses. The goal of the project is to develop a new biofilm mitigation approach using targeted liposomes, which encapsulate an anti-microbial chemical agent, assisted by ultrasound to enhance mixing and attachment rate of the liposomes to the bacterial colonies within the biofilm. In this presentation, we will present some preliminary results of enhancement of penetration of liposomes of nanometer scales into biofilms using mild nonfocused ultrasound. (The spatial average-temporal average power and intensity are 15.1 W and 2.99 W/cm^2, respectively, duty cycle = 50%.)

8:30

3aBAa3. Treatment planning and strategies for acousto-optic guided high-intensity focused ultrasound therapies. Matthew T. Adams (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, adamsm2@bu.edu), Robin O. Cleveland (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), and Ronald A. Roy (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Real-time acousto-optic (AO) sensing has been shown to non-invasively detect changes in *ex vivo* tissue optical properties during high-intensity focused ultrasound (HIFU) exposures. The technique is particularly appropriate for monitoring non-cavitating lesions that offer minimal acoustic contrast. This work employs a modeling-based approach to improve the AO sensing of lesion formation during HIFU therapy, to develop treatment strategies for the ablation of large volumes, and to assess the technique's viability and robustness in a clinical setting. The angular spectrum method is used to model the acoustic field from the HIFU source. Spatio-temporal temperature elevations induced by the absorption of ultrasound are modeled using a finite-difference time-domain solution to the Pennes bioheat equation. Changes in tissue optical properties are calculated using a thermal dose model, calibrated using experimental data. The diffuse optical field is modeled using an open-source GPU-accelerated Monte Carlo algorithm. The Monte Carlo algorithm is modified to account for light-sound interactions, using the acoustic field from the angular spectrum method, and to account for AO signal detection. AO signals are presented in the context of a photorefractive-crystal-based detection scheme, and are compared to signals obtained using standard optical detectors. [Work supported in part by the Whitaker International Program.]

8:45

3aBAa4. Using high-frequency ultrasound to detect cytoskeletal dysfunction in Alzheimer's disease. Ashley N. Calder, Janice E. Sugiyama (Biology, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, ashleycalder09@gmail.com), Laurel A. Thompson (Chemistry, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Recent work indicates that Alzheimer's disease (AD) affects the cytoskeleton and cellular structure through mutations that alter structural proteins, and that dysfunction of the cytoskeleton may play a pivotal role in AD and other neurodegenerative diseases. The goal of our research is to determine if high-frequency ultrasound can detect cytoskeletal dysfunction in AD. Research on the molecular subtypes of breast cancer indicate that mutations specific to each subtype may change the characteristics of the cytoskeleton and resulting properties of the cell such as size, shape, and stiffness. Both computer simulation and experiment have demonstrated that high-frequency ultrasound in the 10–100 MHz range is sensitive to these properties. For this study, ultrasonic tests were conducted on monolayer cell cultures of breast cancer cell lines of different subtypes. The ultrasonic spectra were

compared and correlated to model results using a pattern recognition algorithm. Preliminary results indicate that cell stiffness and size can be determined from the measurements. The cytoskeletal properties of the cells were additionally modified by chemical and physical agents such as the introduction of colchicine and electric fields to mimic the effects of AD. Results from these and future studies with neuron cell cultures will be discussed.

9:00

3aBAa5. Molecular subtyping of colorectal cancer using high-frequency ultrasound. Alexis M. Holman (Biology, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, mgn.alexis@gmail.com) and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Previous studies have shown high-frequency (HF) ultrasound (10–100 MHz) may be sufficiently sensitive to detect and differentiate between both the histopathology and molecular subtypes of breast cancer. In order to explore other uses of HF ultrasound in real-time cancer diagnoses, two colorectal carcinoma cell lines were investigated: HT-29 and SW-620. This experiment tested the sensitivity of HF ultrasound on HT-29 and SW-620 and its ability to differentiate between the two lines. Subtypes of colorectal carcinoma have not been defined, but the morphology of these cell lines indicates that they may possess different molecular subtypes. Cell lines were grown as monolayer cultures and promptly tested with HF ultrasound using a single-element (50 MHz, 6.35-mm) ultrasonic immersion transducer. Ultrasound techniques and conditions were similar to previous breast cancer monolayer tests. Preliminary results indicate the cell lines produce different waveform and spectral signatures. The ability to differentiate between the two cell lines will both broaden the application of HF ultrasonic testing, and provide a method to define colorectal carcinoma subtypes. More cell lines will be tested in an effort to clearly define molecular subtypes of colorectal carcinoma. Defining subtypes will allow for more personalized diagnoses and treatment options for colon cancer patients.

9:15

3aBAa6. Acoustic power delivery for sub-millimeter dimension and deeply implanted medical devices. Marcus J. Weber, Jayant Charthad, Ting Chia Chang, and Amin Arbabian (Elec. Eng., Stanford Univ., 420 Via Palou Mall, Stanford, CA 94305, mjweber3@stanford.edu)

We propose to use acoustic power delivery for sub-mm^{3}medical implants, and we have designed acoustic receivers for investigating this technique of wireless power transmission. Compared with radio frequency and inductive transfer, acoustic power transfer gives more favorable impedance and available power for miniaturized and deeply implanted medical devices. Sub-mm^{3}dimensions are attractive for many different medical applications; however, there is little literature describing the properties or capabilities of miniaturized acoustic receivers, which is the topic of our investigation. Design of efficient miniaturized implants is challenging because of increased losses due to tissue absorption and coupling to parasitic resonant modes. We will present power transfer and impedance measurements of volumetrically scaled acoustic transducers through several thicknesses of tissue. Our measurements show significant available power with high output voltage, resulting from large transducer impedance, which is useful for overcoming threshold voltages of rectifier circuits. Preliminary measurements show the delivery of 340 μW of average AC power to a 1 mm^2 receiver through 3 cm of tissue with an intensity well below the FDA limit. In addition, we will compare our measurements with piezoelectric theory and discuss trends and limits of available power and impedance as a function of volume and transmit distance.

9:30

3aBAa7. Measurement of ultrasonic tissue characteristics of malignant colon cancer cells. Guðfríður Björg Möller, Madilena Mendiola (Phys., Mount Holyoke College, 50 College Ave., South Hadley, MA 01075, gudfridurb@gmail.com), Aislinn Daniels, Dalton Johnson, Rodoula Kyvelou-Kokkaliaris, Kenzi Watkins (Phys., Earlham College, Richmond, IN), and Maria-Teresa Herd (Phys., Mount Holyoke College, South Hadley, MA)

Colon cancer is the third most common cause of death by cancer resulting in over 50 000 deaths a year. The most common method for the detection of colon cancer, a colonoscopy, is quite invasive, and although non-invasive methods are available, they have not proven to be as effective. There have been successful studies on using ultrasound as a diagnostic tool for colon disease in rats. This suggests that ultrasound may be able to be a viable clinical tool for detecting colon cancer and gastrointestinal disease. Here, we explore using quantitative ultrasound to differentiate between benign and malignant colon cells. We made measurements of tissue characteristics of malignant colon cell pellets at different cellular densities. Using ultrasound frequencies ranging from 5 to 20 MHz, we measured the speed of sound, attenuation, and the backscatter coefficients (BSCs). Here, we present the results for the ultrasonic properties of cancerous colon cells and compare these for different densities of the cells.

9:45

3aBAa8. Cough monitoring for pulmonary tuberculosis using combined microphone/accelerometer measurements. Jingqi Fan (ECE, Tufts Univ., 161 College Ave., Medford, MA, jingqi.fan@tufts.edu), German Comina (Laboratorio de Ingeniería Física, Universidad Nacional de Ingeniería, Rimac, Peru), Robert Gilman (Dept. of Int.. Health, Johns Hopkins Bloomberg School of Public Health, Baltimore, MD), Jose Lopez (Unidad de Epidemiología, Hospital Nacional Dos de Mayo, Lima, Peru), and Brian H. Tracey (ECE, Tufts Univ., Medford, MA)

A laboratory-free test for assessing recovery from pulmonary tuberculosis (TB) would be very helpful in regions of the world where laboratory facilities are lacking. Our hypothesis is that analysis of cough sound recordings may provide such a test, as recovering patients should cough less frequently. We have carried out several studies on cough data from a cohort of TB patients in Lima, Peru [Larson *et al.*, PLOS One]. Our previous work provides a foundation to support larger-scale studies of coughing rates over time for TB patients undergoing treatment, but it only used recordings from lapel microphones. For the current study, we use an additional channel recorded by a throat-mounted accelerometer. The accelerometer only responds to patient-generated noise events and thus provides robustness to background noise in the environment. We describe algorithm development for cough data analysis using combined microphone/accelerometer measurements, and compare several event detection and classification strategies. We show that adding the accelerometer improves performance on detection and classification.

10:00

3aBAa9. Quantitative non-linear ultrasonic imaging of targets with high acoustic impedance contrast—Application to bone imaging. Régine Guillermin, Philippe Lasaygues, and Guy Rabau (Ondes et Imagerie, LMA/CNRS, 31, chemin Joseph Aigiuer, Marseille Cedex 20 13402, France, guillermin@lma.cnrs-mrs.fr)

This study focuses on the ultrasonic imaging of high impedance acoustical contrast targets. The aim is to obtain information about shape, dimensions, and sound speed profile of the studied objects. One domain of application is the characterization of long bones. Quantitative information about the acoustic properties of bones tissues are of great interest for diagnosing or treatment monitoring of bone diseases. Inverse scattering problems of this kind are non-linear and various approximations can be used to linearize the scattering equations. Classical methods based on the first-order Born approximation give good results for weakly scattering targets but fail when it comes to give a quantitative information especially for high impedance contrast targets such as bones. In the inversion algorithm proposed here, Green's theorem is used to obtain a domain integral representation of the scattered field. An iterative non-linear algorithm minimizing the discrepancy between the measured and computed scattered fields is used to reconstruct the sound speed profile in the region of interest. The minimization process is performed using a conjugated-gradient method. An experimental study was performed with targets made of paraffin and with lamb bones. Images of the sound speed profile obtained by inversion of experimental data are presented and discussed in both cases.

Session 3aBAb

Biomedical Acoustics: Biomedical Acoustics Best Paper Award Poster Session

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

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

All entries will be on display and all authors will be at their posters from 10:30 a.m. to 12:00 noon.

laBAl0. Comparative analysis of small versus large transducers for high-frequency ultrasonic testing of breast cancer. Student author: **Madison J. Peterson**

laBA7. Improving ultrasound-based estimates of vector displacement fields in elastography applications. Student author: **Sanjay S. Yengul**

laBA8. Boundary conditions in quantitative elastic modulus imaging. Student author: **Daniel T. Seidl**

laBA9. Boundary condition-free elastic modulus reconstructions from ultrasound measured quasi-static displacements. Student author: **Olalekan A. Babaniyi**

lpBAlO. High-frequency ultrasonic measurement of vascularization in phantoms and Avastin-treated mice with breast tumors. Student author: **Andrea N. Quiroz**

lpBA6. Reproducibility of high-frequency ultrasonic signals in breast cancer detection. Student author: **A. Mackay Breivik**

lpBA7. Utah Valley University/Huntsman Cancer Institute Collaborative Breast Cancer Study: High-frequency ultrasound for margin assessments. Student author: **Andrew Chappell**

lpBA9. Ultrasonic phenotyping of breast cancer cells for molecular subtyping. Student author: **Laurel A. Thompson**

2aBA2. Passive mapping of acoustic sources within the human skull cavity with a hemispherical sparse array using computed tomography-based aberration corrections. Student author: **Ryan M. Jones**

3aBAa3. Treatment planning and strategies for acousto-optic guided high-intensity focused ultrasound therapies. Student author: **Matthew T. Adams**

3aBAa4. Using high-frequency ultrasound to detect cytoskeletal dysfunction in Alzheimer's disease. Student author: **Ashley N. Calder**

3aBAa5. Molecular subtyping of colorectal cancer using high-frequency ultrasound. Student author: **Alexis M. Holman**

3aBAa6. Acoustic power delivery for sub-millimeter dimension and deeply-implanted medical devices. Student author: **Marcus J. Weber**

3aBAa8. Cough monitoring for pulmonary tuberculosis using combined microphone/accelerometer measurements. Student author: **Jingqi Fan**

3pBAl. Effects of ambient pressure variation on the subharmonic response from contrast microbubbles. Student author **Nlma Mobadersany**

3pBA2. Ambient pressure estimation using subharmonic emissions from contrast microbubbles. Student author: **Krishna N. Kumar**

3pBA3. Acoustic characterization of polymer-encapsulated microbubbles with different shell-thickness-to-radius ratios using in vitro attenuation and scattering: Comparison between different rheological models. Student author: **Lang Xia**

3pBA4. Nonlinear intravascular ultrasound contrast imaging with a modified clinical system. Student author: **Himanshu Shekhar**

3pBA6. StemBells: Localized stem cell delivery using targeted microbubbles and acoustic radiation force. Student author: **Tom Kokhuis**

3pBA8. Estimation of damping coefficient based on the impulse response of echogenic liposomes. Student author: **Jason L. Raymond**

3pBA9. The stable nonlinear acoustic response of free-floating lipid-coated microbubbles. Student author: **Ying Luan**

4pBAa2. Evaluation of sub-micron, ultrasound-responsive particles as a drug delivery strategy. Student author: **Rachel Myers**

4pBAa4. Response to ultrasound of two types of lipid-coated microbubbles observed with a high-speed optical camera. Student author: **Tom van Rooij**

4pBAa6. Acoustic levitation of gels: A proof-of-concept for thromboelastography. Student author: **Nate Gruver**

4pBAa7. Numerical simulations of ultrasound-lung interaction. Student author: **Brandon Patterson**

4pBAbl0. Surface roughness and air bubble effects on high-frequency ultrasonic measurements of tissue. Student author: **Percy D. Segura**

4pBAb4. Can quantitative synthetic aperture vascular elastography predict the stress distribution within the fibrous cap non-invasively. Student author: **Steven J. Huntzicker**

4pBAb5. Super wideband quantitative ultrasound imaging for trabecular bone with novel wideband single crystal transducer and frequency sweep measurement. Student author: **Liangjun Lin**

4pBAb8. Modeling ultrasonic scattering from high-concentration cell pellet biophantoms using polydisperse structure functions. Student author: **Aiguo Han**

4pBAb9. Characterizing collagen microstructure using high frequency ultrasound. Student author: **Karla P. Mercado**

WEDNESDAY MORNING, 7 MAY 2014 OMNI NARRAGANSETT A/B, 10:00 A.M. TO 12:00 NOON

Session 3aED

Hands-On Acoustics Demonstrations for Middle- and High-School Students

Andrew C. Morrison, Cochair
Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431

Cameron T. Vongsawad, Cochair
Phys. & Astronomy, Brigham Young Univ., 1041 E. Briar Ave., Provo, UT 84604

Acoustiacs has a long and rich history of physical demonstrations of fundamental (and not so fundamental) acoustics principles and phenomena. In this session "Hands-On" demonstrations will be set-up for a group of middle school students from the Providence area. The goal is to foster curiosity and excitement in science and acoustics at this critical stage in the students' educational development and is part of the larger "Listen Up" education outreach effort by the ASA. Each station will be manned by an experienced acoustician who will help the students understand the principle being illustrated in each demo. Any acousticians wanting to participate in this fun event should email Andrew C. H. Morrison (amorrison@jjc.edu) or Cameron Vongsawad (cvongsawad@byu.edu).

Session 3aID

Interdisciplinary: Future of Acoustics

Paul D. Schomer, Chair
Schomer and Associates Inc., 2117 Robert Dr., Champaign, IL 61821

Invited Papers

11:00

3aID1. Criteria for the acoustic environment—What should we do? Paul D. Schomer (Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

The questions are many; the answers I have are few, so I may call upon attendees of this session for help. Who should create policy for the acoustic environment? The F AA(aviation)? FHW A(roads)? DOE(energy)? DOD(defense)? or EPA? or DOI(interior/Park Service)? For example, it has been estimated that about 20 to 30% of the population are noise sensitive? Should policy be based on this minority or on the less sensitive majority? Currently, FAA and DOD allege that their criterion, DNL equal to 65 dB results in about 10 percent highly annoyed; the real percentage is 20 to 30%, so their criterion "protects" only the less sensitive. In a National Park, we are finding that hikers on a moderate length hike (about 4 to 10 km; 2 to 5 h) judge the pleasantness of the acoustic environment during the entire hike largely on the presence or absence of anthropogenic sound. Again there are factions to consider. Roughly 1/3 rate the overall pleasantness on the most pleasant portion of the hike, 1/6 rate on the least pleasant (most unpleasant), 1/6 on the average, and 1/3 on the most recent. What do we do? What should ASA do? Standards? WHY?

11:20

3aID2. Noise pollution in the 21st century. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1147, les@nonoise.org)

What will 21st century noise be and sound like? Will this century be noisier than the 20th century? What changes to noise policy and noise control are we likely? This paper explores these questions. The historical context of the last 100 years will be used to examine and understand the possibilities for the next 100 years.

11:40

3aID3. Future directions in psychoacoustic research facilities. Samuel Gordon, Roger EllingsonNCRAR, Veterans Affairs, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, samuel.gordon@va.gov, David BrowningAltman Browning and Co., Portland, OR, Raymond JanischAcoust., Eckel Industries Inc., Cambridge, MA, and Frederick GallunNCRAR, Veterans Affairs, Portland, OR

The future of psychoacoustic research will involve the measurement of individual differences from a broader range of participants including those with physical disabilities. Auditory testing of physically disabled research participants in a fully anechoic environment presents challenges to the people being tested, to the research team, and to the facility. At foremost concern are the risks of personal injury to physically disabled persons while moving them into, and out from, the testing position in the anechoic chamber. Additional risks of injury exist to the researchers who assist with moving subjects into and out of the chamber. These risks need to be mitigated without compromising the acoustical performance of the anechoic chamber. This paper presents the requirements and the implemented design solutions for an 'Americans with Disabilities Act of 1990' compliant anechoic chamber that is currently being used for auditory research at the Department of Veterans Affairs, National Center for Rehabilitative Auditory Research facility in Portland, Oregon. We advocate that others consider accessibility and safety issues in the development of future psychoacoustic research facilities as we believe such factors are essential to the future of human testing and physical sciences.

Session 3aNS

Noise, Structural Acoustics and Vibration, and ASA Committee on Standards: Wind Turbine Noise

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

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

Kuangcheng Wu, Cochair
Newport News Shipbuilding, 202 Schembri Dr., Yorktown, VA 23693

Invited Papers

8:15

3aNS1. Public complaints about wind turbine noise and adverse health impacts justified. Stephen E. Ambrose (SE Ambrose & Assoc., 15 Great Falls Rd., Windham, ME 04062, seaa@myfairpoint.net), Robert W. Rand (Rand Acoustics, Boulder, CO), Richard R. James (E-Coustics Solutions, Okemos, MI), and Michael A. Nissenbaum (Medical Practice, Rort Kent, ME)

Significant proportions of IWT facility neighbors complain about turbine noise and sleep disturbances, among other adverse health complaints. We undertook an independent evaluation of several wind turbine projects located in Maine, Massachusetts, Vermont, New York, Illinois, Michigan, West Virginia, and Wisconsin to assess if common etiological factors exist. Adverse effects appear to relate to a basket of common factors that were overlooked or not included in preconstruction planning including noise predictions and assessments of likely community reactions. Correcting oversights in future projects should result in quieter IWT projects with reduced or no adverse community reactions. A unified methodology for doing so, enabling wind turbine developers, governmental agencies, municipal boards, and private citizens to assess for potential adverse noise impacts during the permitting phase is presented. Our results are consistent with prior USEPA studies, WHO assessments, and Pedersen and Waye research, among others.

8:35

3aNS2. Approximately 20 Hz plus harmonics amplitude modulated acoustic emissions from a 1.6 MW wind turbine, measurements versus predictions. Kevin A. Dooley (N/A, Toronto, ON, Canada) and Andy Metelka (N/A, 13652 Fourth Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

A recently presented hypothesis and model relating to the generation of spinning modes from wind turbines, as a direct result of acoustic interaction involving the tower, also predicts ~20 Hz plus harmonics low frequency amplitude modulated acoustic emissions as a side effect of the acoustic interaction. The low frequency sound is expected to propagate at measurable amplitudes to the far field (1 km to 2 km). Measurements focused on the ~20 Hz amplitude modulated fundamental and harmonics made at different angles relative to the rotor plane at very close range, and at greater distances are presented. The measurements are compared to predictions based on the tower acoustic interaction hypothesis.

8:55

3aNS3. Examination of the predictions versus measurements of an acoustic interaction model for spinning modes, and concurrent low frequency amplitude modulation acoustic signatures from a 3MW wind turbine. Kevin A. Dooley (N/A, N/A, 55-1817 Harbour Square, Toronto, ON M5J 2L1, Canada, kadooleyinc@rogers.com), Kristy Hansen, and Branko Zajamsek (Mech. Eng., Univ. of Adelaide, Adelaide, SA, Australia)

A recently presented hypothesis and model relating to the generation of spinning modes from wind turbines, as a direct result of acoustic interaction involving the tower, results in a far field infrasound sound pressure level prediction, which is higher than that predicted by point source method. The model also predicts a significant attenuation of the fundamental blade passing frequency component relative to the second and higher harmonics. The model concurrently predicts a low frequency (~20 Hz), amplitude modulated harmonic series as a side effect of the acoustic interaction on a 1.6 MW 80 m diameter wind turbine. This study examines the model predictions of a 3.0 MW 90 m diameter wind turbine, and compares the predictions to measurements of the low frequency harmonic series and blade passing frequency harmonics at several different distances from the wind turbine.

9:15

3aNS4. A comprehensive water tunnel test of a horizontal axis marine hydrokinetic turbine for model validation and verification. Arnie A. Fontaine, Ted G. Bagwell, Michael L. Johnson (Appl. Res. Lab., Penn State Univ., State College, PA), and Dean Capone (Appl. Res. Lab., Penn State Univ., PO Box 30, Sate College, PA 16803, dec5@psu.edu)

As interest in waterpower technologies has increased over the last few years, there has been a growing need for a public database of measured data for these devices. This would provide a basic understanding of the technology and means to validate analytic and numerical models. Through collaboration between Sandia National Laboratories, Penn State University Applied Research Laboratory, and University of California, Davis, a new marine hydrokinetic turbine rotor was designed, fabricated at 1:8.7-scale, and experimentally tested to provide an open platform and dataset for further study and development. The water tunnel test of this three-bladed, horizontal-axis rotor recorded power production, blade loading, near-wake characterization, cavitation effects, and noise generation. Additionally, preliminary comparisons are mode from unsteady CFD for the flow fields measured.

9:35

3aNS5. Equivalent sources method for supersonic intensity of arbitrarily shaped geometries. Nicolas P. Valdivia (Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, nicolas.valdivia@nrl.navy.mil)

Supersonic acoustic intensity is utilized to locate radiating regions on a complex vibrating structure. The supersonic intensity is obtained by a special process that removes the subsonic waves from the near-field acoustical holography measurement. The filtering process is well understood for separable geometries, but unfortunately, there are few results for arbitrarily shaped objects. This work proposes a methodology based on a stable invertible representation of the radiated power operator. The power operator is approximated numerically by the equivalent source formulation and the appropriate complete spectral basis is employed to form the stable invertible operator. The operator is formed with the most efficient radiation modes and these modes are utilized to obtain the supersonic solution for the near-field holographic problem. This concept is tested using numerically generated data in a spherical geometry and the results are validated with the spherical harmonic, supersonic filter. Finally, a vibrating ship-hull structure provides a physical example for application and validation of the proposed methodology in a more complex geometry. [This work was supported by the Office of Naval Research.]

Contributed Papers

9:55

3aNS6. Characterization of noise from an isolated intermediate-sized wind turbine. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

Community-based wind power companies provide subscriptions to individual homeowners and businesses for power generated by a locally installed turbine. Typically, such turbines are of an intermediate size, such as the Vestas V20 120-kW turbines operated by the Cascade Community Wind Company in several locations within Washington state. This model turbine has a tower height of 80 feet with a rotor diameter of 60 ft. Each turbine is installed individually on leased land, with no other turbines nearby. Noise measurements of a turbine located in Thorp, WA, were obtained in a variety of weather conditions. On several occasions with low to moderate wind speeds, the turbine was stopped, enabling the calculation of noise due to the turbine only. Results will be presented showing spectral content and sound pressure level contours for a range of wind speeds.

10:10

3aNS7. Investigations on psychoacoustical and non-acoustical moderators for annoyance evoked by wind turbine noise. Leonid Schmidt (Audio Commun. group, Tech. Univ. Berlin, Blüch4erplatz 14, Aachen 52068, Germany, leonid.schmidt@gmx.net) and André Fiebig (HEAD Acoustics, Herzogenrath, Germany)

In 2012, in total around 23 000 wind turbines are installed on land in Germany. These wind turbines are important to achieve a change in Germany's current energy policy gaining more energy from renewable resources. However, current research underlines that the noise of wind turbines causes noise annoyance and provoke complaints. The sound characteristics of wind turbines depend on many different variables, e.g., type of wind turbine and wind speed. Unfortunately, only little is known about the specific noise characteristics, which are mainly responsible for noise annoyance. Therefore, laboratory experiments are carried out to identify the most annoying noise characteristics of wind turbine noise. The laboratory experiment includes an evaluation of different sounds from wind turbines, manipulated wind turbine sounds, and sounds from other noise sources. The work intends to improve the understanding about the role of psychoacoustic parameters going beyond equivalent continuous sound level. Additionally, the relevance of non-acoustical factors for annoyance caused by wind turbine noise is investigated by interviewing extensively the test subjects.

Session 3aPA

Physical Acoustics: Acoustical Methods and Sensors for Challenging Environments

Dipen N. Sinha, Cochair
Sensors & Electrochemical Devices, Los Alamos Natl. Lab., MPA-11 D429, PO Box 1663, Los Alamos, NM 87545

Cristian Pantea, Cochair
MPA-11, Los Alamos Natl. Lab., MS D429, Los Alamos, NM 87545

Chair's Introduction—7:55

Invited Papers

8:00

3aPA1. Acoustic measurements of rock formations in oilfield boreholes. David L. Johnson (Schlumberger-Doll Res., One Hampshire St., Cambridge, MA 02139, johnson10@slb.com)

Once a borehole is drilled into a rock formation as a potential oil or gas well the environment needs to be characterized by a variety of physical measurements so that e.g. one may know at what depths the hydrocarbon, if any, is located. In this talk I will outline some techniques for measuring, *in situ*, the compressional and shear speeds of sound in a rock formation as a function of depth. Here the "difficult and challenging conditions" are that the measuring instrument is inside the sample (rather than the other way around), the temperatures may reach 175 °C, and the pressure in the borehole may reach as high as 1000 atmospheres.

8:20

3aPA2. Acceleration of acoustical emission precursors preceding failure in sheared granular material. Paul A. Johnson (Geophys., LANL, MS D443, Los Alamos, NM 87545, paj@lanl.gov)

Earthquake precursor observations are becoming progressively more widespread as instrumentation improves, in particular, for interplate earthquakes (e.g., Bouchon *et al*., Nature Geosci., 2013). One question regarding precursor behavior is whether or not they are due to a triggering cascade where one precursor triggers the next, or if they are independent events resulting from slow slip. We investigate this topic in order to characterize the physics of precursors, by applying laboratory experiments of sheared granular media in a bi-axial configuration. We sheared layers of glass beads under applied normal loads of 2–8 MPa, shearing rates of 5–10 μm/s at room temperature and humidity. We show that above ~3 MPa load, precursors are manifest by an exponential increase in time of the acoustic emission (AE), with an additional acceleration of event rate leading to the primary stick-slip failure event. The recorded AE are clearly correlated with small drops in shear stress during slow slip prior to the main stick-slip failure. Event precursors take place where the material is still modestly dilating, yet while the macroscopic frictional strength is no longer increasing. The precursors are of order 100× smaller in recorded strain amplitude than the stick-slip events. We are currently working on statistical methods to determine whether or not the precursors are triggered cascades. [Bouchon *et al*., Nature Geosci. **6**, 299–302 (2013).]

8:40

3aPA3. Using nonlinear ultrasound to measure microstructural changes due to radiation damage in steel. Laurence Jacobs, Kathryn Matlack, Jin-Yeon Kim (Mech. Eng., Georgia Tech, COE Georgia Tech, 225 North Ave. Tech Tower, Atlanta, GA 30332-0360, laurence.jacobs@coe.gatech.edu), Jianmin Qu (civil Eng., Northwestern Univ., Evanston, IL), and Wall J. Joe (EPRI, Charlotte, NC)

The planned life extension of nuclear reactors throughout the United States and abroad will cause reactor vessel and internals materials to be exposed to more neutron irradiation than was originally intended. A nondestructive evaluation (NDE) method to monitor radiation damage would enable safe and cost-effective continued operation of nuclear reactors. Nonlinear ultrasound is an NDE technique that is sensitive to microstructural changes in metallic materials, such as dislocations, precipitates, and their combinations, which are quantified by the measurable acoustic nonlinearity parameter. Recent research has shown the sensitivity of the acoustic nonlinearity parameter to increasing neutron fluence in representative Reactor Pressure Vessel (RPV) steels. The current work considers nonlinear ultrasonic experiments conducted on similar RPV steel samples that had a combination of irradiation, annealing, re-irradiation, and/or re-annealing to a total neutron fluence of 0.5–5 x 1019 n/cm^2(E > 1 MeV) at an irradiation temperature of 290°C. The acoustic nonlinearity parameter generally increased with increasing neutron fluence, and consistently decreased from the irradiated to the annealed state over different levels of neutron fluence. This comprehensive set of results illustrates the dependence of the measured acoustic nonlinearity parameter on neutron fluence, material composition, irradiation temperature, and annealing.

9:00

3aPA4. Materials and fabrication techniques for resonant ultrasound spectroscopy at high and low temperatures. Albert Migliori (NSEC-NHMFL, Los Alamos Natl. Lab., MS E536 Los Alamos, NM 87545, migliori@lanl.gov)

Measurement of the mechanical resonances of materials of interest to condensed matter science is becoming increasingly common because it reveals important thermodynamic signatures such as phase transitions, as well as providing sound speeds and stiffness information for technology. Often done using Resonant Ultrasound Spectroscopy, the ultimate precision of measurements is determined by the mechanical Q, not unusually 104 or higher, thereby making it possible to determine changes in elastic moduli at the sub pert-per-million level. However, resonances and changes in the acoustic response of the cell that holds transducers and the specimen to be measured can introduce artifacts as temperatures change, clouding otherwise important observations. We describe here solutions to such problems with acoustically "dead" materials capable of operation from below 1 K to 900 K using easily available starting components. We also describe strategies for electrical contacts at temperature above the melting point of lead-tin solder. Some unusual results are presented.

9:20

3aPA5. Harsh environment sensors: Aircraft and oil field applications of electromagnetic and acoustic technologies. Edward R. Furlong (General Electric, 1100 Technol. Park Dr., Billerica, MA 01821, ted.furlong@ge.com)

There is an old saying that what can be measured can be improved. However, measuring the key parameters in aircraft engines that drive efficiency and emissions is very difficult. This is even more the case in oil field applications, where extremely high pressures and temperatures are commonly encountered. But the benefits to society of reduced emissions (carbon, pollutants, and noise) and improved oil and gas recovery are tremendous. The task for instrument manufacturers is to develop sensors that are both reliable and cost effective. Recent advances in the application of electromagnetic and acoustic technologies are described for measuring temperature, pressure, flow, and composition in harsh environments. The sensors range from inductively and optically coupled ceramic and silicon devices to ultrasonic and SAW devices packaged with high temperature electronics to fiber optic systems. These sensors are now being deployed on new airframes, engines, deep water wells, horizontal tight shale formations, and high temperature/high pressure wells.

9:40

3aPA6. A resonance technique for the acoustic characterization of liquids in harsh environments. Blake Sturtevant (Los Alamos National Lab., PO Box 1663, MS D429, Los Alamos, NM 87545, bsturtev@lanl.gov)

Accurate knowledge of a liquid's acoustic properties, such as sound speed as a function of temperature and pressure, is important for both basic and applied science. For basic science, sound speed is important for constraining thermodynamic equations-of-state as well as determining elastic nonlinearity parameters. From an applied perspective, sound speed can be used together with other properties to monitor fluid temperature, pressure, and composition. There are important applications, such as in oil/gas or geothermal well characterization, where it is desirable to measure sound speed in liquids in well bore under high pressure, high temperature, and in corrosive environments. However, few experimental sound speeds have been previously reported above 100°C even in liquids as common as water. This talk focuses on the development of a portable, rugged, resonance-based measurement cell for high precision (better than 0.1%) *in-situ* measurements of sound speed in high temperature, high pressure, and corrosive liquids. As an example of the technique, experimentally determined sound speeds in liquid water up to 250°C and 3000 psi will be presented. Acoustic nonlinearity in water, as determined from sound speed as a function of temperature and pressure, will also be discussed.

10:00–10:15 Break

Contributed Papers

10:15

3aPA7. Development, qualification, and performance validation of an optical differential pressure sensor for downhole permanent monitoring applications. James R. Dunphy (Reservoir Monitoring, Weatherford, Wallingford, CT), Omer H. Unalmis (In-Well Flow, Weatherford, 22001 North Park Dr., Kingwood, TX 77339, haldun.unalmis@weatherford.com), and Domino Taverner (Reservoir Monitoring, Weatherford, Wallingford, CT)

This paper describes the development, qualification, and performance validation of an optical differential pressure (DP) sensor for the high-pressure and high-temperature downhole environment. The current implementation of measuring DP in downhole applications is based on the calculated difference between two static pressure sensor measurements with high uncertainties. A true DP sensor is superior in comparison due to the decreased uncertainty both in measurement and component levels. This translates to better performance in system level measurements such as flow. A development program was launched in 2007 to build an optical DP sensor targeted for two main applications: standalone use in single-phase or auxiliary use in multiphase flow measurement systems. Several prototypes have gone through multiple design phases and qualification test programs including mechanical shock and vibration tests, thermal tests, hot vibration tests, short-term stability and long-term endurance tests. The sensor was then integrated with a Venturi and tested in a single-phase flow loop to validate its performance against an electronic DP sensor and an electromagnetic flowmeter. Results suggest that optical DP sensor performs better than the electronic DP sensor in measuring differential pressures, and is on a par with the electromagnetic flowmeter.

10:30

3aPA8. Thermoacoustic engines as self-powered sensors within a nuclear reactor. Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Appl. Res. Lab, P. O. Box 30, State College, PA 16804-0030, sxg185@psu.edu), Randall A. Ali (Elec. Eng., Univ. of the West Indies, Port-of-Spain, Trinidad and Tobago), and James A. Smith (Fundamental Fuel Properties, Idaho National Lab., Idaho Falls, ID)

The core of a nuclear reactor is a particularly harsh environment when functioning properly. When there is an "incident," possibly with the loss of electrical service that accompanied the earthquake and tsunami that struck the Fukushimi Daiichi reactors on 11 March 2011, the term "harsh" seems too tame. We review the development and testing of very simple standing--wave thermoacoustic engines that can be configured as nuclear fuel rods to exploit the temperature differences within that environment rather than try to shield the sensor from "harshness" [U.S. Pat. Appl. Serial No. 13/968,936 (Aug. 16, 2013)]. Those engines produce high amplitude sound that couples to the surrounding heat-transfer fluid to telemeter the information (as frequency and amplitude) to the exterior of the reactor vessel, again without requiring external electrical power. Laboratory results will demonstrate measurement of coolant temperature, identify evolved gases, and provide information about changes in porosity of solids. Thermoacoustic resonances are maintained without use of either an explicit physical hot or cold heat exchanger—a ceramic "stack" is the only required component. We suggest extensions of this approach to other processes that generate substantial temperature gradients, such as industrial crucibles for melting glasses and metals. [Work supported by the U.S. Department of Energy.]

10:45

3aPA9. Nuclear material identification using resonant ultrasound spectroscopy. Cristian Pantea, Tarik A. Saleh, Albert Migliori, Jonathan B. Betts, Erik P. Luther, and Darrin B. Byler (Los Alamos National Lab., MS D429, Los Alamos, NM 87545, pantea@lanl.gov)

Resonant ultrasound spectroscopy (RUS) is a well-established method for determination of the full tensor of elastic moduli of a solid sample in a single frequency sweep. The elastic moduli, together with density, can provide information related to materials fabrication processes, providing a unique signature, or fingerprint, of a material. The goal of this study was to provide forensics for nuclear materials in solid ceramic or metallic form, including composition. The premise of this study was that it is really difficult to find two materials whose density and shear and /or bulk modulus match. We used RUS to determine the bulk and shear modulus for a total of 27 samples. The samples consisted of depleted uranium oxide (MOX) with different doping of Ce, Pu, and Nd oxides, and different methods of fabrication. They were in form of cylinders with flat and parallel faces. Two-dimensional and three-dimensional spaces were investigated, using shear modulus, bulk modulus, and density as variables. The densities varied between 9.0 and 10.6 g/cc, while the shear modulus was 55–80 GPa, with a bulk modulus of 150–240 GPa. The results obtained suggest that there is a good correlation between the elastic moduli and density for samples of different compositions/origins.

11:00

3aPA10. Resonance ultrasound spectroscopy measurements of sandstone at high temperature. Eric S. Davis, Blake T. Sturtevant, Dipen N. Sinha, and Cristian Pantea (Los Alamos Natl. Lab., MPA-11, MS D429, Los Alamos, NM 87545, e.s.davis@tcu.edu)

Deep underground wells, such as those of interest to the oil and gas as well as geothermal industries, are often found in large sandstone formations. In order for drilling, enhancement, and advanced engineering techniques such as hydraulic fracturing to be efficient and successful, the mechanical properties of materials that make up the reservoir must be accurately known. We have used resonant ultrasound spectroscopy (RUS) to determine the physical properties of Berea Sandstone, such as the elastic moduli. In contrast to single crystals or high quality polycrystalline samples, the porous and attenuating nature of sandstone makes an acoustic study of sandstone very challenging. Additionally, the sandstones must be studied at high temperatures in order to simulate conditions that are found in the field. We will present our work on the temperature dependence of the elastic moduli of sandstone (between room temperature and 205 °C.) Our measurements show that Berea sandstone is a very soft material with a bulk modulus of about 6 GPa as compared to 76 GPa for aluminum. Furthermore, a ~10% softening was observed with decrease in temperature, down to a temperature of 110°C, followed by a ~7% hardening down to ambient temperature.

11:15

3aPA11. The propagation of sound above and within a hardbacked rigid porous layer. Hongdan Tao, Bao N. Tong, and Kai Ming Li (Mech. Eng., Purdue Univ., 140 South Martin Jischke, West Lafayette, IN 47907-2031, mmkmli@purdue.edu)

The present paper examines, theoretically and experimentally, the sound field in the vicinity of a non-locally medium due to an airborne source. The non-locally reacting medium is characterized by a porous layer of finite thickness which is placed on a perfectly reflecting plane. According to an asymptotic analysis, the total sound field within the rigid porous medium consists of two components. Each of these two components can be represented by an integral expression. They can then be evaluated by a standard saddle path method to obtain a uniform asymptotic solution. Numerical validation with wave-based numerical schemes demonstrates the accuracy and computational efficiency of the derived asymptotic formula. Additional validation is provided through indoor experimental data obtained by using a layer of glass beads for modeling the rigid porous medium. When the receiver is situated within the porous layer near the perfectly reflecting plane, experimental measurements and theoretical predictions suggest that the interaction of the refracted wave with the perfectly reflecting plane has a significant impact on the total sound field. Experimental data and numerical simulations also indicate that it is rather difficult to distinguish the results between a thin rigid porous layer and a semi-infinite rigid porous medium for an airborne receiver.

Session 3aPPa

Psychological and Physiological Acoustics: Diagnostics of the Pathological Middle Ear by Wideband Acoustic Impedance/Reflectance Measures

Jont B. Allen, Chair
2061 Beckman Inst, 405. N. Mathews, Urbana, IL 61801

Chair's Introduction—8:00

Invited Papers

8:05

3aPPa1. Non-invasive methods for diagnosing diseases of the ear: Wideband acoustic immittance and umbo velocity. Gabrielle R. Merchant (Speech & Hearing BioSci. & Technol., Eaton-Peabody Lab., Harvard-MIT Div. of Health Sci. and Technol., Massachusetts Eye and Ear Infirmary, EPL, Massachusetts Eye & Ear Infirmary, 243 Charles St., Boston, MA 02114, gmerchan@MIT.EDU), John J. Rosowski and Hideko H. Nakajima (Dept. Otology & Laryngology, Eaton-Peabody Lab., Harvard Med. School, Massachusetts Eye and Ear Infirmary, Boston, MA)

Measurements of ossicular motion using laser Doppler vibrometry (LDV) in patients with various ear diseases, along with fresh cadaveric experiments, have increased our understanding of how sound is transduced to the cochlea and how this transduction is modified by various conductive pathologies. These studies have dispelled some misguided beliefs, changed clinical treatments, and shown the potential of LDV as a diagnostic. LDV, however, has substantial limitations as a clinical tool. For patients with conductive hearing loss of unknown etiology (where general otologic exam and conventional tympanometry are not diagnostic), another non-invasive measurement, wideband acoustic immittance (WAI, directly related to power reflectance), in conjunction with audiometric measurements, can differentiate between ossicular fixation, ossicular discontinuity and superior canal dehiscence (SCD). Furthermore, WAI measurements show a common pattern in power reflectance in patients with SCD, with or without a conductive hearing loss. An algorithm to identify this pattern in power reflectance suggests that WAI may be a simple, inexpensive screening tool for SCD. Evidence will be presented from our studies on patients with various otologic diseases, as well as from fresh cadaveric preparations simulating various diseases. Power reflectance can assist in treatment decisions and prevent unnecessary and inappropriate treatments and surgeries.

8:25

3aPPa2. Identifying otosclerosis with battery of aural acoustical tests of absorbance, group delay, reflex threshold, and chirp-evoked otoacoustic emissions. Douglas H. Keefe, Kelly L. Archer (Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68131, Douglas.Keefe@boystown.org), Kendra K. Schmid (Dept. of Biostatistics, College of Public Health Masters Programs, Nebraska Medical College, Omaha, Oregon), Denis F. Fitzpatrick (Boys Town Natl. Res. Hospital, Omaha, NE), M. Patrick Feeney (Natl. Ctr. for Rehabilitative Auditory Res., Veterans Administration and Oregon Health & Sci. Univ., Portland, OR), and Lisa L. Hunter (Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH)

This study evaluated the clinical utility in diagnosing otosclerosis with aural acoustical tests of absorbance, acoustic reflex threshold (ART), and otoacoustic emissions (OAEs) in 23 normal-hearing (NH) ears, 12 ears diagnosed with otosclerosis (OS), and 13 ears after surgical intervention (SU) for otosclerosis. Subjects received audiometric evaluations, and tests of ipsilateral/contralateral ART, pressure reflectance (0.25–8 kHz) parameterized by absorbance and group delay at ambient pressure and at swept tympanometric pressures, and chirp-evoked OAEs (1–8 kHz). ARTs were measured using tonal and broadband noise activators, based on differences in wideband absorbed sound power before and after activator presentation. For NH compared to OS ears, mean ambient absorbance was larger at 4 kHz; mean tympanometric absorbance had larger peak-to-tail differences at low and high frequencies. Probe-to-eardrum length was estimated using group delay at the frequency of the minimum absorbance above 2 kHz, and combined with acoustically estimated area to calculate wideband compensated admittance at the eardrum. Absorbance and compensated group delay revealed complementary information on middle-ear function. Typical OS and SU ear tests showed absent TEOAEs and ARTs, reduced absorbance in OS ears, and anomalous reflectance <1 kHz in SU ears. Other middle-ear conditions showed different patterns of test-battery responses. [Research supported by NIH.]

8:45

3aPPa3. Identification of conductive hearing loss in infants using maximum likelihood analysis of wideband acoustic absorbance and admittance. Beth Prieve and Hammam AlMakadma (Commun. Sci. and Disord., Syracuse Univ., 621 Skytop Rd., Syracuse, NY 13244, baprieve@syr.edu)

Wideband acoustic absorbance (WAA)/reflectance measures of the middle ear identify conductive hearing loss (CHL) in infants and children with excellent accuracy. Recent literature has indicated that WAA analyzed using maximum likelihood ratios in children with conductive hearing loss were more accurate than the clinical standard of single-frequency tympanometry at one frequency. In infants,

single-frequency tympanometry is as effective in identifying conductive hearing loss as wideband acoustic reflectance in one frequency band. The question that arises is whether identification of conductive hearing loss in infants is different than that from children, or, if multiple variable analysis of WAA contributes to the highly significant outcomes. The current project used maximum likelihood ratios to analyze both WAA and admittance measured through tympanometry using three probe tone frequencies. WAA and tympanometry identified CHL equally well in infants, and WAA results are similar to those reported for children. The results suggest that including several frequencies to measure conductive properties of the outer and middle ear is more powerful than single frequencies or bands.

9:05

3aPPa4. Comparisons of reflectance measurements across measurements sessions, instruments, and ages. Susan E. Voss, Defne Abur, Hiwot Kassaye (Eng., Smith College, 100 Green St., Northampton, MA 01063, svoss@smith.edu), and Nicholas J. Horton (Mathematics, Amherst College, Amherst, MA)

Wideband acoustic immitance measures (WAI) are an active area of research aimed at the development of an objective and noninvasive audiometric test that can reliably identify a range of middle-ear disorders. This work compares repeated WAI measurements (absorbance and its closely related quantity power reflectance, in particular) made on normal hearing subjects. In particular, measurements were made on the left and right ears of nine subjects ages 19–22 and seven subjects ages 41 to 47. Each subject returned for repeated measurement sessions between four and eight times. At each measurement session, WAI was measured using two distinct FDA-approved devices: HearID from Mimosa Acoustics and Titan from Interacoustics. This presentation will compare the WAI measurements between two distinct age groups, the two distinct instruments, and across measurement sessions. Additional analyses will be presented to determine if assumptions about ear-canal areas might explain differences between the systems from Mimosa Acoustics and Interacoustics.

9:25

3aPPa5. Estimating the residual ear canal contribution to complex acoustic reflectance measurements using pole-zero fitting. Sarah Robinson (Elec. Eng., Univ. of Illinois at Urbana-Champaign, 2137 Beckman Inst., MC 251, 405 N Mathews Ave., Urbana, IL 61801, srrobin2@illinois.edu), Suzanne Thompson (Commun. Sci. and Disord., St. John's Univ., Queens, NY), and Jont Allen (Elec. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

For diagnostic acoustic measurements of the middle ear, the residual ear canal (REC) between the probe and tympanic membrane (TM) is a significant source of non-pathological variability. Tympanometry measures the TM compliance as a function of canal static pressure, at a single frequency (226 Hz). Alternatively, wideband reflectance is measured at ambient pressure, over a large frequency range (0.2–6.0 kHz). To account for the REC effect, tympanometry assumes that the compliance tends to zero at large static pressures, which may not be a valid assumption (Rabinowitz, 1981), whereas reflectance assumes that the REC contributes a lossless delay. Previously, the authors developed a method to parameterize complex reflectance measurements using pole-zero fits, which may be factored into all-pass and minimum-phase components. The lossless all-pass component approximates the unknown REC delay, while the low-frequency TM compliance may be estimated from the minimum-phase component. Using this approach, we evaluate middle ear static pressure data from a controlled study of cadaver ears (Voss, 2012) and an *in vivo* study in which subjects were trained to induce negative middle ear pressure, as well as controlled syringe measurements. Our results indicate that the TM compliance is not zero at the static pressure extremes measured under tympanometry.

9:45

3aPPa6. Development of a finite element model for normal and pathological middle ears: Impedance, reflectance, and sweep frequency impedance. Sunil Puria, Hongxue Cai (Mech. Eng., Stanford Univ., 496 Lomita Mall, Stanford, CA 94305, puria@stanford.edu), Shinji Hamanishi (Mech. Eng., Sendai National College of Technol., Miyagi, Japan), Kevin N. O'Connor, and Charles Steele (Mech. Eng., Stanford Univ., Stanford, CA)

A 3D 'virtual middle ear model' using finite-element modeling techniques was developed to simulate the dynamics of the human middle ear. COMSOLMultiphysics software was used to solve the resulting acoustics-structure interaction problem. The model is validated by comparing numerical results with experimental data measured in the ear canal (EC), on the tympanic membrane (TM), umbo, stapes footplate, and cochlear pressure. The model consists of anatomy from μCT imaging and material parameters from the literature (Cai *et al*., PLOS One, in review). The EC impedance Zec, reflectance Rec, and the pressure Pec due to a constant displacement Dec (Wada *et al*., 1998) were calculated for the normal middle ear, with the stapes blocked, and with the stapes disarticulated. The results in this virtual model are consistent with experiments performed in both cadaveric and living ears. The model is used to analyze the sensitivity and specificity of Zec, Rec, and Pec due to variations in the material properties of the middle ear including the TM, ossicles, malleus-incus, and incus-stapes joints, and the footplate. [Work supported in part by grant R01-DC05960 from the NIDCD of NIH and by a Fellowship from the Institute of National Colleges of Technology, Japan.]

Contributed Papers

10:05

3aPPa7. Cochlear reflectance: Measurements and modeling. Daniel Rasetshwane and Stephen T. Neely (Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68131, daniel.rasetshwane@boystown.org)

Cochlear reflectance (CR), the cochlear contribution to ear-canal reflectance (ECR), has theoretical advantages for cochlear modeling. Comparisons between measurements and models may lead to improved interpretation of cochlear status and provide a basis for making improvements to the models. Previous evaluation of clinical utility of CR measurements showed that CR did not predict auditory status or behavioral threshold as accurately as other otoacoustic emission measurements. Strategies for improving the quality of CR measurements were assessed in ECR measurements from 27 participants. Results indicate that the quality of CR measurements can be improved by (1) increased averaging time and (2) adjustment to the methods for extracting CR from ECR. Simulation of ECR was performed using a combination of a middle-ear model and a one-dimensional cochlear model. Simulated CR was the ECR difference between active and passive conditions of the model. The model simulation results were compared with measurements of ECR and CR in both the time and frequency domains. Minor disparities between measurements and model will provide a basis for improvements in the model. Substantial agreement between measurements and model suggest that CR is consistent with linear coherent reflection due to random impedance perturbations along the cochlear partition.

10:20

3aPPa8. Acoustic horn reflectance: Equations and measurements. Stephen T. Neely and Daniel Rasetshwane (Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68131, stephen.neely@boystown.org)

Reflectance is the transfer function between forward and reflected components of pressure waves that propagate in wave guides such as acoustic horns. Exact solutions to Webster's Horn Equation are only known for a few specific shapes, including parabolic, conical, and exponential. Explicit equations for reflectance in these three horn shapes were recently published for infinite-length horns. Measured reflectance in 3D-printed, finite-length examples of these horn shapes, show no similarity in the frequency-domain to exact reflectance for infinite-length horns. The similarity improves after adjustments to both the equations and the measurements. New equations were derived for exact reflectance of finite-length horns. Measured reflectance was smoothed by time-domain windowing. In contrast to frequency-domain reflectance, comparisons of time-domain reflectance prior to the time sound reaches the end of the horn were not much affected by these adjustments. Because exact equations are known and 3D-printed examples are easy to obtain, these three horn shapes may be useful as standards for comparing different reflectance-measurement systems.

WEDNESDAY MORNING, 7 MAY 2014 BALLROOM A, 8:00 A.M. TO 11:00 A.M.

Session 3aPPb

Psychological and Physiological Acoustics: Binaural Processing and Spatial Perception (Poster Session)

Jayaganesh Swaminathan, Chair
Boston Univ., 635 Commonwealth Ave., Rm. 320, Boston, MA 02215

All posters will be on display from 8:00 a.m. to 11:00 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 9.30 a.m. and contributors of even-numbered papers will be at their posters from 9:30 a.m. to 11:00 a.m.

Contributed Papers

3aPPb1. Investigating stream segregation and spatial hearing using event-related brain responses. Le Wang (CompNet, Boston Univ., 677 Beacon St., Boston, MA 02215, lwang@bu.edu), Samantha Messier (Biomedical Eng., Boston Univ., Boston, MA), Scott Bressler (CompNet, Boston Univ., Boston, MA), Elyse Sussman (Neurosci., Albert Einstein College of Medicine , Bronx, NY), and Barbara Shinn-Cunningham (CompNet, Boston Univ., Boston, MA)

Several studies have used auditory mismatch negativity (MMN) to study auditory stream segregation. Few of these studies, however, focused on the stream segregation that involves spatial hearing. The present study used MMNs to examine the spatial aspect of stream segregation. Traditional oddball streams were presented in a passive listening paradigm, either in isolation or in the presence of an interfering stream. The interfering streams were engineered so that the deviants were not unexpected if the two streams were heard as perceptually integrated. Interfering streams were either spectrally distant from or close to the oddball stream, and were also spatially separated from the oddball stream. The deviant stimuli differed from the standards in perceived spatial location. For comparison, the MMN paradigm developed by Lepistö *et al.* (2009) using intensity deviants was repeated on the same group of subjects. For both paradigms, the MMN was strongest when the oddball stream was presented in isolation, less strong but present when the two streams were spectrally separated, and not observable when the streams were spectrally close. These results demonstrate the feasibility of using the MMN to measure spatial stream segregation, especially in populations for whom task-based behavioral experiments cannot be undertaken.

3aPPb2. Spatial influences on change detection within complex auditory scenes. Kelly Dickerson (Army Res. Lab., 131 Waldon Rd., Abingdon, MD 21009, dickersonkelly23@gmail.com), Jeremy Gaston (Army Res. Lab., Aberdeen, Massachusetts), and Angelique Scharine (Army Res. Lab., Aberdeen, MD)

Change deafness is the auditory analog to change blindness. Both phenomena represent a tendency to miss large changes in the environment, suggesting that sensory experiences are not verbatim and may details crucial for detection and identification. Spatial separation facilitates change detection in vision, but its role in auditory change detection is unclear. In this study, we examined the impact of spatial separation on the detection of appearing and or disappearing sound sources in an auditory scene. Participants listened to a brief auditory scene (1000 ms) comprised of four sources followed by a scene where a sound source was added or subtracted from the scene or where no change occurred. There were two listening conditions, where the sound sources were each distributed across a loudspeaker array, or the sound sources were all played from a single loudspeaker. Results indicate that listeners were better able to detect appearing than disappearing sounds, and fewer errors were made when sound sources were spatially separated. These results are consistent with an attention-based explanation of change detection failures. Further, the beneficial effect of spatial separation suggests that change blindness and deafness may share a common mechanism.

3aPPb3. Spatial attention in an auditory dual task. Nandini Iyer (Air Force Res. Lab., 2610 Seventh St., Bldg. 441, Area B, Wright Patterson Air Force Base, OH 45433, Nandini.Iyer@wpafb.af.mil), Eric R. Thompson (Ball Aerosp., Dayton, OH), Griffin D. Romigh (Air Force Res. Lab., Dayton, OH), Carryl L. Baldwin (George Mason Univ., Fairfax, VA), and Brian D. Simpson (Air Force Res. Lab., Dayton, OH)

Research on divided auditory attention has focused on the ability of listeners to report keywords from two spatially separated simultaneous talkers (Best *et al*., 2006); however, the information that listeners extract from each talker is the same (i.e., keyword identification). In realistic listening environments, there is often a competing demand for auditory attention; listeners might be required to monitor critical auditory events while also responding to an ongoing auditory signal. It is not clear how spatial separation affects performance in auditory dual-tasks under varying levels of task difficulty. Listeners identified an ongoing stream of color/number keywords originating at 0° azimuth (primary task), while detecting the presence of a critical call sign originating from locations ranging from -45° to $+45^\circ$ (secondary task). The difficulty of the primary task was varied by introducing noise or by requiring stimuli identification in an auditory one- or two-back memory recall task. The difficulty of the secondary task was varied by increasing the set-size of critical call signs that listeners had to monitor and by changing the SNRs. Preliminary results indicate that a listener's ability to detect the presence of a critical call sign increased as its spatial location moved further away from 0° azimuth and performance was modulated by the difficulty of the primary task. The result has important implications for spatial attention as a function of task difficulty in multitalker environments.

3aPPb4. Mislateralization of sound images at dichotically presented a rippled noise signal and masker. Olga Milekhina and Alexander Supin (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru)

A mislateralization effect was observed when a target signal and a masker were presented dichotically: the target signal in one ear and the masker in the other ear. The target signal was band-filtered rippled noise with periodical interchange of ripple peaks and troughs positions. This interchange produced periodical timbre variations in the sound image. The masker was non-rippled band-filtered noise. In 100% of the trials, the image of periodical variations of timbre reflecting interchanges of the ripple peaks and troughs in the target signal was perceived (released from masking) at low ripple density and signal/masker ratios down to -35 dB. However, lateralization of the image depended on the signal/masker ratio. The image was correctly lateralized toward the ear of the target signal when the signal was of a higher level than the masker. The image was wrongly lateralized toward the ear of the masker presentation (mislateralized) when the masker was of a higher level than the signal. Implications to mechanisms of dichotic release from masking are discussed: the release is possible without the spatial-attention mechanism.

3aPPb5. Effects of cognitive load on selective and divided auditory spatial attention. Daniel McCloy (Inst. for Learning and Brain Sci., Univ. of Washington, Box 357988, Seattle, WA 98115-7988, drmccloy@uw.edu) and Adrian KC Lee (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

In a previous experiment [McCloy and Lee, 2013, *J. Acoust. Soc. Am.* **134**, 4230 (2013)], we reported an asymmetry between phonetic and semantic detection tasks with respect to spatially separated vs. spatially adjacent attended word streams: phonetic tasks showed high false alarm rates when to-be-attended streams were spatially divided, while semantic tasks did not. In this experiment, we manipulate the difficulty of the semantic task to investigate the effect of cognitive load on task performance. We compare trials in which the two to-be-attended streams comprise words drawn from either one or two semantic categories, and from categories with either a small or large number of words. Effects of these manipulations on target hit rate and false alarm rate are discussed in relation to previous work.

3aPPb6. A detection theoretical framework for conceptualizing the bottom-up and top-down processes during concurrent-source segregation. Yi Shen (Dept. of Cognit. Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697, shen.yi@uci.edu)

During a behavioral task involving the segregation of two concurrent sound sources, bottom-up and top-down processes could both influence the task performance. Therefore, the task performance alone is not assured to provide a complete description of perceptual segregation and does not allow the inference of the contributions from the bottom-up and top-down mechanisms. In the current study, a modeling framework for the perceptual segregation of concurrent harmonic complexes was proposed. The modeling framework allowed the experimenter to probe the status of the segregation, including (1) whether concurrent harmonic complexes were perceived as segregated sources, (2) which one of the segregated sources was used to perform the experimental task, and (3) whether the listener was able to correctly identify the perceived sources. In two experiments, listeners detected changes in the spectral shape of a target harmonic complex when a masker complex was simultaneously presented. By fitting the proposed model to behavioral data, the status of perceptual segregation, usually hidden from the experimenter, could be revealed. Furthermore, the proposed model was able to quantitatively predict the effects of fundamental frequency differences and target-to-masker ratio on concurrent profile analysis using a small number of interpretable parameters.

3aPPb7. Rapid binaural processing for source segregation and lateralization. Darrin K. Reed, Angela Josupeit, and Steven van de Par (Univ. of Oldenburg, Achternstrasse 23, Oldenburg 26122, Germany, darrinreed@hotmail.com)

For realistic listening conditions, interaural cues will fluctuate due to the presence of multiple active sources. If it is assumed that the binaural system is sluggish, then the perceived location of the sound input would be an average of the varying interaural cues. If, however, the binaural system is fast enough to assess the rapidly changing interaural differences, then it could be possible for the binaural system to properly identify the spatial position of a target source. Using a continuous, broadband noise stimulus that contained periodically alternating interaural time differences (ITD) and, notably, no monaural cues, we investigated the binaural system's ability to lateralize brief durations of the target ITD. Results show that listeners can lateralize targets for durations of 3–6 ms indicating that the binaural system allows for a segregation and lateralization of the target and interfering noise streams. Furthermore, results indicate that the binaural system mediates the buildup of a modulated stream. A second experiment investigating whether the salience of the target ITD in the aforementioned stimulus depends on the temporal position of the target within the phase of an amplitude modulated envelope revealed that this was not the case.

3aPPb8. Binaural masking release: An increase in workload capacity. Jennifer Lentz (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, jjlentz@indiana.edu) and James Townsend (Dept. of Psychol. and Brain Sci., Indiana Univ., Bloomington, IN)

The following study applied reaction time analyses of workload capacity to tone-in-noise detection for monaural (NmSm), diotic (tone and noise identical at each ears; NoSo), and dichotic (tone is anti-phase but noise is not; NoSᴛᴛ) conditions. Reaction times allow comparisons between these conditions at the same signal-to-noise ratios (something which cannot often be done using threshold and percent correct) and can also expose dynamic contributions to the release from masking provided by binaural interactions. Here, we apply a reaction-time based capacity coefficient, which provides an index of workload efficiency. We demonstrate that the release from masking generated by the addition of an identical stimulus to one ear (NmSm vs. NoSo) is unlimited capacity (efficiency $\approx$1), consistent with an independent parallel-channel model. However, the release from masking generated by the anti-phasic tone (NoSo vs. NoSᴛᴛ) leads to a significant increase in workload capacity (increased efficiency)—most specifically at lower signal-to-noise ratios. These experimental results provide further evidence that configural processing plays a critical role in binaural masking release, and that these mechanisms operate more strongly when the signal stimulus is difficult to detect.

3aPPb9. Cortical neural correlates of the binaural masking level difference (BMLD). Heather J. Gilbert, Trevor M. Shackleton, Katrin Krumbholz, and Alan R. Palmer (MRC Inst. of Hearing Res., University Park, Nottingham NG7 2RD, United Kingdom, trevor@ihr.mrc.ac.uk)

Single-cell responses to binaural masking level difference (BMLD) stimuli were measured in the primary auditory cortex of Urethane-anesthetised guinea pigs. Firing rate was measured as a function of the presentation level of 500 Hz S0 and Sπ pure tone signals in the presence of N0 and Nπ maskers. The maskers were white noise, low-pass filtered at 5 kHz, with a spectrum level of 23 dB SPL. Responses were similar to those previously reported in the inferior colliculus (IC). At the lowest tone signal levels, the response was dominated by the noise masker, at higher signal levels the firing rate either increased or decreased. Signal detection theory was used to determine detection threshold. Very few neurones yielded measurable detection thresholds for all four stimulus conditions, and there was a wide range in thresholds. However, across the entire population, the lowest thresholds were consistent with human psychophysical BMLDs. Tone and noise delay functions could be used to predict the shape of the firing-rate vs. signal-level function. In summary, like in the IC, the responses were consistent with a cross-correlation model of BMLD with detection facilitated by either a decrease or increase in firing rate.

3aPPb10. Binaural masking level difference on hearing impairments compensating by hearing aids. Cheng-Yu Ho (Dept. of Biomedical Eng., School of Biomedical Sci. and Eng., National Yang-Ming Univ., No.155, Sec. 2, Linong St., Beitou Dist., Taipei City 11221, Taiwan, swellfishyu@gmail.com), Shuenn-Tsong Young (Holistic Education Ctr., Mackay Medical College, New Taipei City, Taiwan), Wen-Ying Yeh, and Zhi-Hong Wang (Dept. of Otolarngology-Head and Neck Surgery, Tri-Service General Hospital, Taipei City, Taiwan)

Many researchers reported that the Binaural masking level differences are reduced in pathological conditions, such as sensorineural hearing loss, retro-cochlear hearing loss, central nervous system disease, and so on. In former studies revealed that the amount of BMLDs are about 2–3 dB or in minus. However, for sensorineural hearing loss, the adaptive method may be limited by the hearing thresholds of subjects. Therefore, to eliminate limitation of hearing thresholds, we delivered stimuli by a pair of sound chambers and receiving by hearing aids fitting with NAL-NL1 formula through headphones to subjects. There were eight mild to moderate sensorineural hearing loss subjects (6 females and 2 males, average 62.2 y/o, std. 9.7) participating in this study. With white noise as masker, we conducted pure tone detection thresholds for 125, 250, 500, 1000 and 2000 Hz in S0N0 and SπN0 conditions. The pilot results showed that the threshold differences (S0N0—SπN0) are 4, 6.3, 4.1, 6.9, and 6.3 dB from 125 to 2k Hz. The pilot results revealed that there are BMLDs on SNHLs with compensating by hearing aids. That is to say, this testing method may be used to rule out supra-threshold deficits from sensorineural hearing losses.

3aPPb11. Extent of lateralization caused by interaural time differences of high-frequency click-trains. Regina M. Baumgaertel and Mathias Dietz (Dept. of Medical Phys. and Acoust. and Cluster of Excellence 'Hearing4All', Oldenburg Univ., Carl-von-Ossietzky Str. 9-11, Oldenburg 26129, Germany, regina.baumgaertel@uni-oldenburg.de)

Hafter and Dye [J. Acoust. Soc. Am. **73**, 644 (1983)] measured threshold interaural time differences (ITDs) for high-frequency click-trains. They found that, given a fixed duration, threshold ITDs were better (lower) for lower pulse rates. Studies by other authors measured lateralization for broad band click-trains and were generally not focused on how the low-frequency fine-structure ITD may influence the results. Such an influence may become prominent at ITDs > 600 μs when fine-structure ITDs alone are subject to a cue reversal. The current study therefore focused on high-frequency click-trains and a broad range of ITDs (up to at least 2 ms). The extent of lateralization elicited by these click-trains was determined using an acoustic pointer procedure. Interclick intervals of 5, 10, 20, and 50 ms were investigated. Subjects show an increase in lateralization with increasing ITD even when exceeding the physiologically plausible range of 600 μs. Interclick intervals of 5 ms generally cause the lowest extent of lateralization, in line with the threshold ITD data. The results will be discussed in the light of spatial cue enhancement for cochlear implants, assuming a similar extent of lateralization for interaural pulse time differences at low pulse rates.

3aPPb12. Mapping spatial release from informational masking with one or two masker talkers. Eric R. Thompson (Ball Aerosp. & Technologies Corp., 2610 7th St., Bldg. 441, Wright Patterson, OH 45433, eric.thompson.ctr@wpafb.af.mil), Nandini Iyer, Griffin D. Romigh, and Brian D. Simpson (Air Force Res. Labs, Wright-Patterson AFB, OH)

Spatially separating a target talker from a colocated masker can improve intelligibility, an effect referred to as spatial release from masking (SRM). In this study, listeners' ability to identify speech was measured using the Coordinate Response Measure (CRM) corpus with one or two same-sex CRM maskers as a function of the position of the maskers and the target-to-masker ratio. The target and maskers were each filtered into non-overlapping bands to reduce energetic masking. The target was always straight ahead of the listener, and the masker(s) were presented on the horizontal plane at positions from -90° to $+90^\circ$ in azimuth using individualized head-related transfer functions (HRTF). Consistent with prior results, performance is worst when the target and masker(s) are colocated and improves as the distance between target and maskers increases. Also, performance is better when the two maskers are colocated with each other than when they are separated, and is better when the two maskers are both on the same side of the target than when one masker is on each side of the target. The data suggest spatial filtering strategies that listeners may adopt to improve performance in multitalker scenarios.

3aPPb13. The role of amplitude modulation in auditory 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) and Paul W. Anderson (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

The ratio of direct to reverberant sound energy (D/R) has been shown to be a primary acoustic cue to perceived sound source distance. Because it is unclear exactly how D/R might be encoded in the auditory system, a variety of more physiologically plausible correlates to D/R have been identified, including: spectral variance, interaural correlation, and temporal cues. Here, following recent neural work by Kuwada and Kim [ARO (2014)], we describe a new correlate to D/R and perceived distance related to the amplitude modulation (AM) depth of the signal at the listener's location. This cue is caused by the change in the modulation transfer characteristics of the room as a function of source distance. Results from an apparent distance estimation task confirm the efficacy of this AM depth cue in a reverberant soundfield (approximate broadband T_{60} = 3 s), when level cues are made ineffective. Distance estimates were found to be more accurate when the source signal (1-octave band of noise centered at 4 kHz) had AM (32 Hz, 100% depth), and this facilitation was only observed in reverberation. The facilitation was most evident for monaural input, indicating that the AM depth cue is likely processed monaurally.

3aPPb14. Spatial release from masking in musicians and non-musicians. Jayaganesh Swaminathan, Christine R. Mason, Timothy M. Streeter, Gerald Kidd, Jr. (Boston Univ., 635 Commonwealth Ave., Rm. 320, Boston, MA 02215, jswamy@bu.edu), and Aniruddh D. Patel (Tufts Univ., Cambridge, MA)

Recent research suggests that musically trained individuals have enhanced speech-in-noise perception, raising questions about the mechanisms underlying these effects. However, to probe the robustness of this finding and to evaluate theories about the possible mechanisms responsible for this performance advantage, it is desirable to examine speech-in-noise perception using a variety of methods. This study assessed the differences in spatial release from masking (SRM) in musicians and non-musicians. Spatially separating a speech target from interfering masker(s) generally improves target intelligibility; an effect known as spatial release from masking. A speech target was presented simultaneously with two or four speech maskers that were either colocated with the target (0° azimuth) or were symmetrically separated from the target in azimuth (±15° for two maskers; ±15° and ±30° for four maskers). Preliminary results for the two-masker condition indicated greater SRM in musicians than in non-musicians with the differences largely driven by lower target-to-masker ratios for musicians in the spatially separated condition. For the four-masker condition the SRMs observed for the musicians and non-musicians were more similar. However, large individual differences in performance were noted particularly for the non-musically trained group. Future research directions will be discussed, to explore the mechanisms behind these effects. Work supported by NIH-NIDCD and AFOSR.]

3aPPb15. Significance of height loudspeaker positioning for perceived immersive sound field reproduction. Antonios Karampourniotis, Sungyoung Kim (Elec., Comput., TeleCommun. Eng. Technol., College of Appl. Sci. and Technol., Rochester Inst. of Technol., 347 culver Rd., Apt. 5, Rochester, NY 14607, tonykaramp@gmail.com), Doyuen Ko (Audio Eng. Technol., Mike Curb College of Entertainment & Music Business, Belmont Univ., Nashville, TN), Richard King, and Brett Leonard (Sound Recording, Schulich School of Music, McGill Univ., Montreal, QC, Canada)

Recently, new multichannel audio formats incorporating height loudspeakers have caught researchers' attention due to their ability to reproduce an immersive sound field. This study investigated the influence of the height loudspeaker positions and their signals on individually perceived sound quality. The authors generated nine-channel sound sources by convolving two anechoic musical pieces with nine selected room impulse responses measured in different distances and heights. In the listening test two layers of loudspeakers were used: the horizontal layer, following the standard ITU-R BS 775 five-channel loudspeaker configuration and the height layer (with elevation of 30°) with a total of twelve loudspeakers, located at ±30, ±50, ±70, ±90, ±110, and ±130 degrees. Four height signals were reproduced through eight different configurations of four height loudspeakers. Twelve listeners participated in the experiment, wherein they were asked to compare the randomly presented eight configurations and rank them based on their individually perceived sound quality. The experimental results indicate that despite the perceptual differences related to the room impulse responses, the perceived overall quality is significantly influenced by the positioning of the four height loudspeakers.

3aPPb16. Availability of envelope interaural time-difference cues does not improve front/back localization of narrow-band high-frequency targets via head movement. Ewan A. Macpherson (Natl. Ctr. for Audiol., Western Univ., 1201 Western Rd., Elborn College 2262, London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

Information about the front/rear location of a sound source is available in the relationship between the direction of head rotation and the direction of changes in interaural time and level differences (ITD and ILD). Our previous results show that such dynamic cues are highly effective for low-frequency stimuli, but minimally effective for narrowband high-frequency stimuli, in which, respectively, ITD and ILD cues are primarily available. In this study, we assessed the possible benefit for dynamic localization of providing more robust envelope ITD cues in high-frequency stimuli. Listeners judged the front/rear location of anechoic free-field stimuli presented over the central portion of a slow (~0.25 Hz), continual, 90-degree head oscillation. Stimuli were bursts of wideband (0.5–16 kHz), low-frequency (0.5–1 kHz), or high-frequency (6–6.5 kHz) random-phase noise or of raised-sine stimuli with exponent 2, modulation frequency 125 Hz, and bandwidth 6–6.5 kHz. Localization accuracy was high for wideband and lowpass stimuli but poor (and similar) for high-frequency noise and raised-sine stimuli, despite listeners' measured ITD JNDs for raised-sine stimuli being significantly lower than for high-frequency noise. The results suggest that neither veridical dynamic ILD nor ITD cues can overcome the erroneous spectral cue for front/back created by narrowband high-frequency stimuli.

3aPPb17. How high frequency envelopes influence spatial localization in rooms. Salwa Masud, Hari Bharadwaj (Dept. of Biomedical Eng., Boston Univ., 677 Beacon St., Boston, MA 02215, smasud@bu.edu), and Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

Perception of sound laterality (left-right angle) is mediated by both interaural time differences (ITD) and interaural level differences (ILD). Previous localization studies in anechoic settings consistently show that low-frequency ITDs dominate perception of source laterality. However, reverberant energy differentially degrades ITDs and ILDs; the effects of room reflections on the perceptual weight given to ITDs and ILDs are not well understood. Here, we tested the hypothesis that high-frequency envelope ITD cues are important for spatial judgments in reverberant rooms by measuring the perceived laterality of high-pass, low-pass and broadband sounds. Results show that when ILD cues and ITD envelope cues are both available, reverberant energy has the smallest effect on localization of high-pass stimuli. When ILD cues are set to zero, localization of high-pass stimuli with strong envelopes (i.e., click trains and speech tokens) is also minimally affected by reverberant energy; however, as envelope modulation is reduced, subjects show increasing localization bias, responding towards the center. Moreover, for stimuli with strong envelopes, subjects with better modulation detection sensitivity are affected less by the addition of reverberant energy. These results suggest that, in contrast to in anechoic space, high-frequency envelope ITD cues influence localization in reverberant settings.

3aPPb18. Weighting of interaural time difference and interaural level difference cues in wide-band stimuli with varying low and high frequency energy balance. Ewan A. Macpherson (The Natl. Ctr. for Audiol., Western Univ., London, ON, Canada) and Tran M. Nguyen (Health and Rehabilitation Sci. Graduate Program, Western Univ. , 205 Oxford St. east, London, ON, Canada, tnguy45@uwo.ca)

Wideband stimuli carry both interaural time and level difference (ITD and ILD) sound location cues. Previously, listener weighting of those cues has only been measured for low-pass, high-pass, and flat spectrum wide-band conditions [Macpherson and Middlebrooks, JASA (2002)]. In this study, we determined how weighting of ITD and ILD cues varied with the low- and high-frequency energy balance in wide-band stimuli. Listeners reported locations of targets that were presented over headphones using individual head related transfer functions. ITD and ILD cues were manipulated by attenuating or delaying the sound at one ear (by up to 300μs or 10 dB), and the final weight was computed by comparing the listener's localization response bias to the imposed cue bias. Stimuli were 100-ms bursts of noise whose spectra were flat from 0.5 to 2 kHz and from 4 to 16 kHz with a level difference between those low- and high-frequency ranges varying in 10-dB steps from −30 to + 30 dB. ILD weight increased (from ~0.5 to 1.5 deg/dB) with increasing high-frequency energy, but ITD weight was constant (~0.08 deg/us) across spectral profiles. The results suggest that in wideband stimuli, weighting of ILD is more stimulus dependent than weighting of ITD.

3aPPb19. The role of interaural level differences in the localization of low-frequency sine tones. Brad Rakerd (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, rakerd@msu.edu), Zane D. Crawford, and William M. Hartmann (Phys. and Astronomy, Michigan State Univ., East lansing, MI)

Five human listeners reported the azimuthal locations of low-frequency sine tones presented in free field, either by a 180-degree loudspeaker array or by virtual reality. The virtual sources were synthesized using cross-talk cancellation based on signals continuously monitored in the listener's ear canals. The experiment tested the duplex model of sound localization, especially the role of interaural level differences (fixed ILD = 0, 6, or 12 dB) for frequencies of 750 Hz or less. Trials with real sources, baseline virtual sources, and virtual sources with manipulated ILDs were always combined in all experiments. Experiments showed that the interaural time differences (ITD) dominated most fixed opposing ILDs, as previously reported, only when interaural phase differences (IPD) were less than 90 degrees. When IPDs exceeded 90 degrees, the ITD lost its influence, and localization was dominated by ILDs. Localization judgments never followed IPDs across a 180-degree boundary, except when a zeroed ILD caused judgments to become chaotic. Within the 90-degree IPD limit, judgments for fixed ILD appeared to follow the ITD better than the IPD. Abnormal interaural conditions (e.g., ITDs and ILDs of opposite sign) led to a notable increase in front-back confusions. [Work supported by AFOSR grant FA9550-11-1-0101.]

3aPPb20. Binaural interference with dynamic interaural cues. Jacqueline M. Bibee (Speech and Hearing Sci., Univ. of Washington, Seattle, WA) and G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu)

Sensitivity to interaural time and level difference (ITD and ILD) in high-frequency amplitude-modulated sound is reduced by simultaneous presented low-frequency "interferers." Often, the magnitude of this "binaural interference" is limited by the salience of the target cue. This study addressed the hypothesis of reduced interference for salient cues by restricting target cues to sound onset (high salience; condition "R0") or sound offset (low salience for ITD, high salience for ILD; condition "0R"). In control condition "RR," cues remained constant over duration. Targets were trains of 16 Gabor clicks (4 kHz carrier frequency, 2 ms interclick interval). In baseline conditions, targets were presented in isolation; in interference conditions, targets were gated simultaneously with a diotic 500 Hz tone. ITD or ILD detection thresholds were measured adaptively. Results demonstrated significant binaural interference across conditions, despite the use of pulsatile modulators and regardless of the presence or absence of onset cues. Ceiling effects resulted in unmeasurably high ITD (but not ILD) thresholds in some conditions. Consistent with the low salience of envelope ITD near sound offset, this occurred for a majority of subjects when interferers were present but onset cues were not. Work supported by R01-DC011548.]

3aPPb21. Temporal weighting of interaural time differences in low frequency noise presented at low signal-to-noise ratio. Anna C. Diedesch and G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, anna.c.diedesch@vanderbilt.edu)

Discrimination of interaural time difference (ITD) improves with increasing duration of a target stimulus, but more slowly than expected if ITD sensitivity was temporally uniform over the sound duration. Houtgast and Plomp [JASA **44**, 807–812 (1968)] thus argued for nonuniformal temporal weighting of ITD, in which sound onsets dominate listeners' ITD judgments. That theory is well supported by recent work. Additional data reported by Houtgast and Plomp suggest more uniform weighting in the presence of masking noise at 5 dB signal-to-noise ratio (SNR). The current study measured ITD thresholds for 500 Hz octave-band noise targets with ITD fixed over duration (condition "RR") or changing linearly from zero to peak value (condition "0R") or vice versa ("R0"). Targets were presented in the presence or absence of a continuous 500Hz octave-band masker (5dB SNR). Comparison data were obtained using 500 Hz pure-tone targets across identical ITD and masker configurations (Diedesch and Stecker, 2014, Assoc. Res. Otolaryngol.). ITD detection with pure-tone targets did not appear to benefit (in terms of threshold-vs-duration slopes) from masking noise as in Houtgast and Plomp (1968). Other stimulus conditions that more closely replicate the conditions of that study (i.e., noise targets) are discussed. [Work supported by NIH R01 DC011548.]

3aPPb22. The effect of masker spatial uncertainty on sound localization. Brian Simpson (Air Force Res. Lab., 2610 Seventh St., Area B, Bldg. 441, Wright-Patterson AFB, OH 45433, brian.simpson@wpafb.af.mil), Robert Gilkey (Wright State Univ., Dayton, OH), Nandini Iyer (Air Force Res. Lab., Wright-Patterson AFB, OH), Eric Thompson (Ball Aerosp. & Technologies Corp, Fairborn, OH), Griffin Romigh (Air Force Res. Lab., Wright-Patterson AFB, OH), and Douglas Brungart (Walter Reed National Military Medical Ctr., Bethesda, MD)

Previous research from our laboratory has shown that uncertainty about the spatial location of a masking sound (randomly selected from 1 of 239 locations) can dramatically reduce localization accuracy for a simultaneous target relative to the case in which the masker location is known exactly. One possibility is that knowing the masker location enables the listener to establish a spatial attention filter at that location to suppress the masker and better localize the target. In this experiment, the level of masker spatial uncertainty was systematically varied across blocks by varying the number of possible masker locations (1, 2, 4, 8, or 239) and informing subjects about these possible locations prior to the start of each block. Localization errors were found to increase systematically in the left/right, front/back, and up/down dimensions as the number of potential masker locations increased; this effect was most prominent in the left/right dimension, where localization errors increased by nearly 30 degrees across conditions. Moreover, for a masker in a given location, errors generally increased across these levels of masker spatial uncertainty, consistent with the notion that there is a cost to distributing attention across multiple locations.

3aPPb23. Just noticeable difference of source-receiver distances in the auralization process using speech and music signals. Bernardo Murta, Priscila Wunderlich, Jessica J. Lins de Souza, and Stephan Paul (Undergrad. Program in Acoust. Eng., Federal Univ. of Santa Maria, Av. Roraima 1000, Santa Maria, RS 97105900, Brazil, bernardo.murta@eac.ufsm.br)

The precision of source-receiver transfer functions is of importance to provide reliable and ecologically valid results in auralization. Recently the overall just noticeable difference (jnd) in signals obtained from the convolution of music with slightly varying source-receiver transfer functions was determined using a paired comparison procedure (sound are equal/are different) and estimating the jnd from the 75% point on the psychometric curve. JNDs were 3.55 cm for dislocations approaching the source and 3.46 cm if going away from the source. Now we present the results from tests using a speech signal convolved with the same simulated impulsive responses of different source-receiver positions making 3 sets of 23 stimuli pairs. Each one represents receiver dislocations in one dimension: x, y, and z. The jnd using a speech signal was found to be 3.86 cm when approaching the source and 3.97 cm for the opposite direction. The difference calculated between the jnd of the two tests was 0.314 cm (approaching the source) and 0.506 cm (in the opposite direction). Further tests will be done using noise signals and an overall jnd will be computed.

3aPPb24. The lowest signal-to-noise ratio at which the precedence effect operates for signals in noise. Fiona Guy (MRC/CSO Inst. of Hearing Res. - Scottish Section, New Lister Bldg., Glasgow Royal Infirmary, Glasgow, United Kingdom, fiona@ihr.gla.ac.uk) and Michael Akeroyd (MRC/CSO Inst. of Hearing Res. - Scottish Section, Glasgow, Strathclyde, United Kingdom)

The precedence effect is a robust auditory phenomenon which aids localization of sound in a reverberant room. Its primary characteristic is that the perceived location of the sound is based on the first arriving, direct, sound, with subsequent reflections mostly ignored. Most research into the precedence effect uses stimuli presented in quiet. Here, we used signals in noise, and considered what is the lowest signal-to-noise ratio (SNR) at which the precedence effect remains operative. In our experiment, the stimuli were speech or click trains, presented as direct + reflection pairs over headphones. Crucially, the ITDs of the stimuli set were such that, when presented alone, the direct signal was lateralized to one side of the head but the reflection to the other. A two-interval, two-alternative, lateralization task was used to measure performance. If the reflection was truly ignored, then the perceived lateralization of the direct + reflection pair would give the correct answer to a trial, but if the reflection was not ignored, then the perception would give the wrong answer. The task is SNR dependent, allowing a "precedence threshold" to be measured using an adaptive procedure that varied the SNR of the direct + reflection pair in noise. We report results from both normal and hearing-impaired listeners. [Work supported by the Medical Research Council and the Chief Scientist Office, Scotland.]

3aPPb25. Temporally diffusive reflections and the precedence effect. M. Torben Pastore (Architectural Acoust. and Ctr. for Cognition, Commun. and Culture, Rensselaer Polytechnic Inst., 4 Irving Pl., Troy, NY 12180, m.torben.pastore@gmail.com), Jens Blauert (Inst. of Commun. Acoust., Ruhr-Universität , Bochum, Germany), and Jonas Braasch (Architectural Acoust. and Ctr. for Cognition, Commun. and Culture, Rensselaer Polytechnic Inst., Troy, NY)

In reverberant conditions, humans routinely demonstrate an ability to form the auditory event in the direction of the first wavefront and thus identify the direction of the (physical) sound source—the so-called "Precedence Effect." Within limits, this effect even holds when the reflected sounds (the lag) contain more energy than the direct sound (the lead). Previously, we investigated the lateral extent of the Precedence Effect for specular reflections. The current research extends this inquiry by investigating the effect of temporally diffusive reflections, using the same stimuli, but convolving the lag with a 2-ms Hanning windowed Gaussian noise. The lead and lag stimuli were 200-ms Gaussian noise (500-Hz center frequency, 800-Hz bandwidth) presented dichotically with a programmable amount of temporal overlap. The lag/lead level ratio was increased in 2-dB steps, the lead/lag interval was varied from −5 to 5-ms in steps of 1-ms. Listeners indicated the lateralization of their auditory events with an acoustic pointer. The resulting temporal smearing substantially decorrelates lead and lag while keeping them essentially related. The Precedence Effect is found to be more robust to increases of the lag level for diffusive stimuli than for the previously tested sets of specular reflections.

WEDNESDAY MORNING, 7 MAY 2014 BALLROOM B, 11:00 A.M. TO 12:00 NOON

Session 3aPPc

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Alfred L. Nuttall, Chair
Dept. Otolaryngology, Oregon Health & Sci. Univ.,, 3181 S.W. Sam Jackson Park Rd., Portland, OR 97239

Chair's Introduction—11:00

Invited Paper

11:05

3aPPc1. The role of physics in inner-ear physiology and auditory perception. Egbert de Boer (Audiol., Academic Medical Ctr., Meibergdreef 9, Amsterdam 1105AZ, Netherlands, e.d.boer@hccnet.nl)

Auditory analysis of acoustical stimuli has mainly been connected with Fourier analysis. This touches the basic link between auditory perception and the operation of the hearing organs, namely, physics and mathematics. This relation has amply been demonstrated by the work and ideas of Georg von Békésy. Later, the healthy cochlea (inner ear) was found to contain amplifying elements that boost frequency selectivity. Associated with this there is a pronounced nonlinearity. Furthermore, the ear does not only absorb and process sounds, it also emits sound waves. Mathematical models of all these processes must contain subsets serving them all, a goal that has not yet been reached. On the contrary, recent findings on the movements of the various membranes and cells in the organ of Corti have increased the difficulties of mathematical modeling. Additional subjects to be covered in the lecture are (1) in psychophysics of hearing: pitch perception, inharmonic sounds, critical bands, hearing of patients with hearing loss, and auditory revalidation, and (2) in auditory neuroscience: recording of single fibers of the auditory nerve, reverse correlation, inverse solutions of cochlear mechanics, nonlinear analysis, and optical coherence tomography (OCT).

Session 3aSAa

Structural Acoustics and Vibration, Underwater Acoustics, and Physical Acoustics: Session in Honor of Murray Strasberg

David Feit, Cochair
ASA, 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502

Dean Capone, Cochair
Penn State, PO Box 30, State College, PA

Chair's Introduction—8:25

Invited Papers

8:30

3aSAa1. Murray Strasberg, a role model. David T. Blackstock (Appl. Res. Labs. & Dept. of Mech. Eng., Univ. of Texas at Austin, PO Box 8029, Austin, TX 78713-8029, dtb@austin.utexas.edu)

Murray Strasberg held several different offices in the Acoustical Society: Executive Council Member 1969–1972, President 1974–1975, and Secretary 1987–1990. The Society was very fortunate to have Murray at its helm in 1974–1975. Wallace Waterfall died early in Murray's term. ASA old timers remember Wallace as the man who "ran the Acoustical Society out of his hip pocket" for years and years. How were we going to get along without him? Murray led us painstakingly through the difficult reorganization. Also up was the Standards issue. ASA decided to write and publish its own acoustical standards rather than leave them to other organizations. Finally, the recent birth of INCE had sadly led to turf wars, which had to be dealt with. Several years later, long after most presidents have happily gone out to pasture, Murray again stepped up, in this case to fill the void caused by Betty Goodfriend's retirement as ASA Secretary. Murray served as ASA's last Secretary. Beyond his splendid service to the Acoustical Society, Murray was a renowned acoustical scientist and, perhaps above all, a warm and generous friend.

8:50

3aSAa2. Murray Strasberg—A lifetime dedicated to the Acoustical Society of America. Charles E. Schmid (Acoust. Society of America, 365 Ericksen Ave., Bainbridge Island, Washington 98110, ceschmid@att.net) and Elaine Moran (Acoust. Society of America, Melville, NY)

Murray Strasberg received the Gold Medal in 2000 "for contributions to hydroacoustics, acoustic cavitation and cavitation noise, and for dedicated service to the Society" when he was 83 years old. He should have received it much earlier, but the classified nature of his work prevented the preparation of a complete nomination dossier. In addition, most members did not fully realize how much Murray did for the Society as his contributions were spread over a half-century. While serving as President (1974–1975), the Society was faced a critical emergency when Wallace Waterfall, the Society's mainstay and current treasurer passed away. Murray stepped in and worked with others to handle the financial operations of the Society until a new Treasurer was selected. He also served in many other Society roles including Associate Editor of JASA, chair of the committee that set the groundwork for the Congressional Science and Engineering Fellowship Program, and as Secretary from 1987 to 1990. We will share some personal stories about Murray recalled from the many decades working with him as a colleague and wise counsel.

9:10

3aSAa3. Murray Strasberg's contributions to acoustic cavitation and bubble dynamics. Lawrence." Crum (Appl. Phys. Lab., CIMU, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, lac@apl.washington.edu)

Although Murray Strasberg's most significant contributions to acoustics were classified and not available to the general public, he also published some significant papers in the general area of acoustic cavitation and bubble dynamics. Some of these papers are so fundamental in scope that I consider them required reading for any new graduate student that joins our group. At cavitation sessions in the ASA, Murray was always asking insightful questions that amazed others who did not know of his pioneering work on this topic. In this presentation, I will describe some of Murray's contributions to bubble dynamics and cavitation as well as relate some personal observations of my interactions with him over our respective professional careers.

9:30

3aSAa4. Murray Strasberg's contributions to the Navy's ship silencing programs. David Feit (ASA, 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502, feit.d@att.net)

Murray's long career with the U.S. Navy, extended over seven decades 1942–2012, and began when he joined the David Taylor Model Basin (DTMB), now the Carderock Division of the Naval Surface Warfare Center), in 1942. Soon after joining DTMB he was assigned to making cavitation noise measurements on World War II submarines or with model propellers in water tunnels. This lead to his long standing interest in cavitation and bubble noise, matters to be discussed by others in this session. I will discuss his work in structural acoustics, which grew out of his involvement with a propeller related signature source and ultimately proved to be one of his most significant contributions to the Navy's submarine silencing efforts.

9:50

3aSAa5. Hydrodynamic noise—Murray Strasberg's legacy. William Blake (Adjunct Prof. Dept. Aero. Mech. Eng., Univ. of Notre Dame, 6905 Hillmead Rd., Bethesda, MD 20817, hydroacoustics@aol.com)

The decade of the 1950s provided us with the beginnings of sub-disciplines of acoustics that we now call aeroacoustics and hydroacoustics. In the beginning, the attention was placed on mechanisms relevant to the aeronautical engineering and the early rocket and space vehicle communities. Accordingly, much published work at the time dealt with jet noise and structural fatigue resulting from that noise and from turbulent boundary layer excitation. In 1956, Murray Strasberg and Hugh Fitzpatrick published a seminal paper, "Hydrodynamic Sources of Sound", 1st Hydrodynamics Symposium. This paper put the aerodynamic noise theory of Lighthill (1952) in the context of Navy application and defined relevant source types. A. Prosperetti will discuss Murray's legacy regarding bubble noise and cavitation discussed in that paper. I will discuss the other interest of Murray: i.e., flow induced vibration and sound. Although he published little on this subject the impact that he had on others who did was important. Accordingly, Murray had continuing impact on the developing knowledgebase of flow-induced sound and vibration, and we will use the area of TBL noise as an example of how concepts in flow noise and vibration have evolved under Murray's career span.

10:10–10:30 Break

10:30

3aSAa6. Murray Strasberg and bubble acoustics. Andrea Prosperetti (Mech. Eng., Johns Hopkins Univ., 223 Latrobe Hall, Baltimore, MD 21218, prosperetti@jhu.edu) and Andrea Prosperetti (Appl. Sci., Univ. of Twente, Enschede, Twente, Netherlands)

Murray Strasberg made seminal contributions to the nucleation and acoustics of bubbles. Half a century after publication, these papers still receive a sizable number of citations every year. The talk will review this work, comment on its impact, and put Strasberg's classical results in a modern perspective.

10:50

3aSAa7. Oscillations of nonspherical and spherical bubbles. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu), Thomas J. Asaki (Mathematics Dept., Washington State Univ., Pullman, WA), and David B. Thiessen (Chemical Eng. and Bioeng. Dept., Washington State Univ., Pullman, WA)

Murray Strasberg maintained an interest in bubble oscillations throughout his career and frequently offered an incisive perspective. His early bubble research includes [Strasberg, J. Acoust. Soc. Am. **25**, 536–537 (1952); Strasberg, Acustica **4**, 450 (1954)]. His pertinent application of an electrostatic potential theory analogy expands interests of some of James Clerk Maxwell's students. In addition some related discussions with Murray will be recalled and experiments from the 1980s and 1990s at Washington State University pertaining to bubbles and their associated dynamics reviewed. Those experiments concern the response of bubbles to steady and modulated optical radiation forces [Unger and Marston, J. Acoust. Soc. Am. **83**, 970–975 (1988); Unger and Marston, Ocean Optics IX, Proc. SPIE **925**, 326–333 (1988)] and to modulated acoustical radiation forces [Asaki *et al*., Phys. Rev. Lett. **75**, 2686–2689, (E) 4336 (1995); Asaki and Marston, J. Fluid Mech. **300**, 149–167 (1995); Asaki and Marston, J. Acoust. Soc. Am. **102**, 3372–3377 (1997)]. The latter papers pertain to the simultaneous measurement of the frequency and damping of bubble shape oscillations. [Work supported by ONR.]

11:10

3aSAa8. Low-frequency propeller forces and sound—Murray Strasberg's legacy. Jason Anderson (NSWC Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, jason.m.anderson@navy.mil)

In the early 1960s, Murray Strasberg, after having already made significant contributions to the field of hydroacoustics in the areas of propeller cavitation noise and general hydrodynamic sources of sound, turned his attention toward low-frequency forces that are generated by propellers, which ingest non-uniform flow. Dr. Strasberg developed an elegantly simplistic approach to estimate the propeller unsteady thrust by actually measuring the sound pressure radiation from a prototype propeller installed in a wind tunnel test section. Murray's approach required that the propeller be acoustically compact, thereby allowing the propeller to be treated as a single dipole sound source. This approach offered by Murray, for which a U.S. patent was awarded in 1982, served as a simpler alternative to the more traditional and complicated approach of force dynamometry. Dr. Strasberg focused his efforts on measuring propeller thrust at the blade passage frequency tonals, but he did, however, encourage Maurice Sevik to perform his seminal propeller turbulence ingestion thrust research in the 1960s and 1970s, which truly ushered in the modern era of propeller turbulence ingestion research that continues today. In this presentation, we will describe Murray Strasberg's novel propeller unsteady thrust measurement approach, and we will discuss how researchers over the years have built upon Murray's foundational concept.

Contributed Papers

11:30

3aSAa9. Mass sensing using the time domain response of a functionalized microresonator array. Aldo A. Glean, Joseph F. Vignola, John A. Judge (Mech. Eng., Cathilic Univ. of America, 620 Michigan Ave., N.E, Washington, DC 20064, 10glean@cardinalmail.cua.edu), and Teresa J. Ryan (Mech. Eng., East Carolina Univ., Greenville, NC)

Energy flow between size scales in vibrating structures can be used for mass detection. This work considers structures composed of a larger scale primary resonator (transduction element) and a set of substantially smaller attached resonators (sensing elements). Functionalization of the sensing elements in the downscale realm allows for detection of specific chemical vapors, biological agents, etc. Common approaches for this type of mass detection involve monitoring a structure's frequency domain response for downward shifts in resonance frequencies as mass binds to the sensing elements, or inferring mass changes from shifts in response shapes of the sensing elements. Instead of using frequency domain information, this work describes a detection method based on observing time histories of the energy exchange between the transduction and sensing elements. The energy initially introduced to the system at the transduction element is drawn into the sensing elements. Some energy will return upscale at a later time that depends on the system characteristics. This work demonstrates that by functionalizing every second sensing element, the concentration of adhered mass can be related to the profile of energy returned to the transduction element. Sensor limitations and optimized designs based on measurement noise and fabrication tolerances are also reported.

11:45

3aSAa10. Response shaping using a subordinate oscillator array. John A. Sterling (US Navy, 9500 Macarthur Blvd., Bethesda, MD 20817, john.a.sterling1@navy.mil) and Joseph Vignola (Mech. Eng., Catholic Univ. of America, Silver Spring, MD)

Recent research has shown that arrays of small dynamic elements attached to a master structure can be tuned to significantly alter the time or frequency response of the system. Colloquially known as "fuzzy structures," subordinate oscillators have led to applications including damping, radio frequency filtering, energy harvesting, and micro electro-mechanical system (MEMS) chemical vapor sensing. A passive machinery damping system will be designed and tested for silencing properties. The current subordinate oscillator array (SOA) design consists of a plate of sheet metal with arrays of cantilevers machined of similar but different lengths. These cantilevers will have a range of natural frequencies, which correspond to a desired frequency suppression range. When the SOA is mounted to a vibration source, it functions as an acoustic meta-material which traps and dissipates energy. This is accomplished by synchronizing the phase and frequency of the cantilevers with machinery peak amplitude frequencies. By designing cantilevers properly, the SOA acts as a mechanical broadband filter as opposed to a notch filter. The SOA will be tested primarily for vibration suppression performance but also for sensitivity to tolerance and energy storage density.

WEDNESDAY MORNING, 7 MAY 2014 — 552 A, 8:25 A.M. TO 11:55 A.M.

Session 3aSAb

Structural Acoustics and Vibration and Noise: Environmental Vibration

Calum Sharp, Cochair
Univ. of Salford, The Crescent, Salford M5 4WT, United Kingdom

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

Chair's Introduction—8:25

Invited Papers

8:30

3aSAb1. Draft American National Standard: Methods for measuring the vibration response of the ground. James E. Phillips (Wilson, Ihrig & Assoc., Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

Working Group 14 of the Acoustical Society of America/American National Standards Institute S2 Mechanical Vibration and Shock Committee (ASA/ANSI S2 WG14) was formed to develop a standard on methods for measuring the vibration response of the ground to be used when assessing potential impact upon vibration sensitive receivers in the vicinity of rail transit systems. The contents of the standard will be based upon industry accepted methods as described in the Federal Transit Administration guidance manual "Transit Noise and Vibration Impact Assessment" (FTA-VA-90-1003-06) while considering new developments since the latest version of the guidance manual was published in May 2006. This paper will describe the background of the methods and the structure, content, and status of the draft standard.

8:50

3aSAb2. Predicting indoor groundborne noise and vibration levels from transit sources. Shannon McKenna (ATS Consulting, 215 N Marengo Ave. Ste. 100, Pasadena, CA 91101, smckenna@atsconsulting.com)

The groundborne noise and vibration impact thresholds in the Federal Transit Administration (FTA) Noise and Vibration Impact Assessment guidance manual apply to indoor spaces. Therefore, the prediction methodology must account for how the building structure affects groundborne vibration through floor resonances and coupling loss and how the building affects groundborne noise through sound absorption or radiation. This presentation will include measurement results that illustrate the variation in building response to vibration. Due to the large number of vibration sensitive receivers that may be adjacent to a proposed transit line, it can be difficult to account for the variation in building response throughout the corridor when only a limited number of measurements sites are available or practical. Case studies will be presented that include recommendations for how measurement results can be incorporated into the prediction procedure.

9:10

3aSAb3. Validation of the Pipe in Pipe vibration software model to determine ground-borne noise and vibration levels above construction tunnels and the determination of end corrections for different train operating scenarios. Graham Parry (Environment, ACCON UK Ltd., Unit B, Fronds Park, Frouds Ln., Aldermaston, Reading, Berkshire RG7 4LH, United Kingdom, graham.parry@accon-uk.com), Steve Summers (Acoust., Anderson Acoust., Brighton, United Kingdom), and David Yates (Acoust., ACCON UK Ltd., Reading, United Kingdom)

The London Crossrail project requires that re-radiated noise from the construction of the tunnels should achieve exceptionally stringent criteria with respect to groundborne noise and vibration. The challenge has been to implement robust vibration modeling for the movement of construction trains within the tunnel in order to derive noise levels within sensitive properties (including recording studios) above the tunnel. The Pipe in Pipe (PiP) software model (which is based on elastic continuum theory) developed by Hunt and Hussein has been modified and refined empirically utilizing the measurement of exceptionally low vibration levels emanating from the temporary underground construction railway track. The paper describes the extensive and challenging monitoring and modeling which been carried out. The study represents the first rigorous method of determining noise and vibration at properties above the tunnel from construction trains for the Crossrail project, with an associated detailed validation exercise in line with ISO 14837-1 'Mechanical vibration—Ground-borne noise and vibration arising from rail systems'. A combination of modeling and vibration measurements were used to inform the predictions of groundborne noise levels, which can be adjusted to replicate the behavior of other track and rail support types.

9:30

3aSAb4. The acceptability of railway induced vibration in residential environments. James Woodcock, Eulalia Peris, Gennaro Sica, Calum Sharp, Andy T. Moorhouse, and David C. Waddington (Acoust. Res. Ctr., Univ. of Salford, Newton Bldg., Salford M5 4WT, United Kingdom, j.s.woodcock@salford.ac.uk)

The aim of the study presented in this paper is to investigate the use of self-reported acceptability for assessing the human response to environmental vibration in residential environments. The human response to environmental stressors such as noise and vibration is often expressed in terms of exposure-response relationships that describe annoyance as a function of the magnitude of the vibration. These relationships are often the basis of noise and vibration policy and the setting of limit values. This paper takes a different approach by expressing exposure-response relationships for vibration in terms of self-reported acceptability. It is argued that exposure-response relationships expressing acceptability as a function of vibration exposure will complement existing relationships for annoyance in future policy decisions regarding environmental vibration. The results presented in this paper are derived from data collected through a large scale (N = 1431) socio-vibration survey conducted in the United Kingdom, the aim of which was to derive exposure-response relationships for vibration in residential environments. The sources of vibration considered are railways and construction.

9:50

3aSAb5. Differences in the human response to freight and passenger railway vibration in residential environments. Calum Sharp, James Woodcock, Eulalia Peris, Andrew Moorhouse, and David Waddington (Acoust. Res. Ctr., Univ. of Salford, The Crescent, Salford M5 4WT, United Kingdom, c.sharp@edu.salford.ac.uk)

The aim of this paper is to quantify and investigate differences in the human response to freight and passenger railway environmental vibration. Data for this research comes from a field study comprising interviews with respondents and measurements of their vibration exposure ($N = 752$). A logistic regression model has been developed to classify measured railway vibration signals in the field study as freight or passenger signals, with a classification accuracy of 96%. Exposure-response relationships for annoyance due to exposure to freight and passenger railway vibration are then determined using an ordinal probit model with fixed thresholds. These exposure response relationships indicate that the annoyance response for exposure to freight railway vibration is significantly higher than that for passenger railway vibration. In terms of a community tolerance level, the population studied is 15 dB (re 10^{-6} m s^{-2}) more tolerant to passenger railway vibration than freight railway vibration. The potential reasons for this difference in the human response are investigated and discussed. Some of the factors that are investigated include time of day effects, sleep disturbance, effects of combined noise and vibration and the effects of social, attitudinal, and demographic factors.

10:10–10:30 Break

10:30

3aSAb6. Vibration, noise, and their interacting contributions toward sleep disturbance. Michael G. Smith, Ilona Croy, Oscar Hammar, and Kerstin Persson Waye (Occupational and Environ. Medicine, The Sahlgrenska Acad. at the Univ. of Gothenburg, Box 414, Gothenburg 40530, Sweden, michael.smith@amm.gu.se)

The market share of goods traffic operating on the European rail networks is expected to almost double from 2001 to 2020. Nocturnal time slots are expected to play an important part in facilitating this increase and as such sleep disturbance in residential areas is expected to be the most significant hindering factor. Little data currently exist that may be utilized to investigate the potential impact. Within the European project CargoVibes we experimentally investigated sleep disturbance due to vibration and noise arising from freight trains. An experimental trial was conducted involving 23 healthy subjects sleeping for six nights; a habituation night, a control night, and four nights with combinations of vibration and noise exposure. The primary objective was to examine the contribution of each exposure, and investigate how vibration and noise contribute individually and simultaneously to human response. The secondary aim was to determine whether increased numbers of events relative to previous work using 20–36 trains further impacted on sleep. Physiological parameters including sleep stage, cardiac activity, and cortical arousals were obtained using polysomnography and questionnaires were administered to obtain sleep quality data. Results indicate that vibration directly contributes to sleep fragmentation. The findings shall be further discussed at the conference.

10:50

3aSAb7. Health based approach for regulations for vibrations from rail traffic. Martin van den Berg (Ministry of Env., Rooseboomstraat 69, Den Haag 2593PB, Netherlands, m.vdb@xs4all.nl)

Several countries adopted regulations for vibrations and a few made this even statutory. Most do however not endeavor to regulate vibrations are make weak references to—old—standards. Partly, this seems due to the perceived complexity, partly to a lack of knowledge, and partly to the feeling that vibrations are somehow less of a problem. In this paper, a procedure is described to derive limit values for rail vibrations according to WHO-rules for the use of epidemiological evidence in environmental risk assessment. Recent developments in the EU-project CARGOVIBES made it possible to get sufficient data to make this possible. Not all elements that are necessary for a stable regulatory system are obtained, but at least politicians may be supported much better in the decisions for a better protection of the population. Worrying gaps in knowledge are the influence of night exposure on health, the interaction with noise exposure and the effectiveness of measures.

Contributed Papers

11:10

3aSAb8. Acoustical characterization of grass-covered ground. Chelsea E. Good, Joseph F. Vignola, Aldo A. Glean, John A. Judge (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, 26GOOD@cardinalmail.cua.edu), Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC), Jacob Sunny (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Diego Turo (BioEng., George Mason Univ., Fairfax, VA)

An investigation of acoustical properties of soil covered with live grass is presented. Measurements of normal surface impedance of such samples have been performed over a 200–2000 Hz frequency band using a vertical impedance tube. A phenomenological model is used to predict acoustic properties of the samples. In this work, the samples are considered as a three-component system: soil, grass roots, and foliage. Acoustic impedance of this composite material has components resulting from the different constituent elements. In order to differentiate the acoustic absorption contribution of each element, grass that was controlled for both water content and grass blade height was grown. The acoustic contribution of the soil was determined by performing measurements on unseeded ground with an equivalent watering protocol. Contributions of the roots and foliage were determined by making impedance measurements before and after shearing the mature foliage near the soil surface. The effect of water content in the soil was estimated by making measurements of the samples before and after oven desiccation. We show the effects of roots and foliage on acoustic absorption of grass-covered ground and the acoustic parameters of these complex media estimated using an equivalent fluid model.

11:25

3aSAb9. Prediction and assessment of environmental vibrations from railway operations on Marmaray. Mehmet Caliskan and Salih Alan (Dept. of Mech. Eng., Middle East Tech. Univ., Ankara 06800, Turkey, caliskan@metu.edu.tr)

Marmaray project involves upgrading of commuter lines on both sides of Bosphorus with an uninterrupted, modern, high-capacity commuter rail system. The line totals approximately 76 km including an immersed tunnel under Bosphorus. In this study prediction and assessment of environmental vibration levels due to railway traffic along 20-km portion between Gebze and Pendik on the Asian side are presented. Experimentally obtained existing soil-structure coupling and structural amplification factors for the whole line are applied to the theoretically calculated vibration levels for use in assessment studies. Vibration mitigation measures are devised with respect to Turkish Environmental Noise Regulation and criteria by the Federal Transit Administration.

11:40

3aSAb10. Application magnetorheological elastomer to dynamic vibration absorber for vibration reduction by avariable-unbalance excitation. Unchang Jeong, Jin-Su Kim, Jung-Min Yoon, Jae-Eung Oh (Mech. Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, unchang.jeong@gmail.com)

This paper presents a concept of dynamic vibration absorber (DVA) applied magnetorheological elastomers (MREs) for vibration reduction. Elastic modulus of MRE significantly increases due to the induced magnetic field. Elastic modulus changes the stiffness of DVA. Thus, the DVA can work effectively in a wide frequency range instead of a narrow bandwidth as a conventional dynamic vibration absorber does. Numerical simulations of avariable-unbalance excitation system are used to validate its effectiveness. Thus, the MRE-DVA will be applicable to the vibration reduction.

Session 3aSC

Speech Communication: Topics in Speech Production (Poster Session)

Alexis R. Johns, Chair

Univ. of Connecticut, 406 Babbidge Rd., Unit 1020, Storrs, CT 06269

All posters will be on display from 8:00 a.m. to 12:00 noon, To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10.00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

3aSC1. Temporal characteristics of vowel articulation: Electromyographic investigation of the "articulatory period". Amy LaCross and E. Fiona Bailey (Dept. of Physiol., Univ. of Arizona, 1501 n. Campbell, Rm. 4104, PO Box 245051, Tucson, AZ 85724, lacross@email.arizona.edu)

In speech, articulator movement precedes the acoustic signal [e.g., Meyer (1991); Gracco (1988); Lubker and Gay (1982); Bell-Berti and Harris (1981)]. The onset and offset of movement, i.e., the 'articulatory period' (Bell-Berti and Harris, 1981, p. 13) is said to encompass the acoustic signal however the precise timing and variability of this period are not well understood. Using electromyography (EMG), we examine the onset/offset of the articulatory period as a function of phonetic context. We recorded EMG activity from the posterior and anterior regions of the genioglossus (GG) muscle of the tongue in subjects during the articulation of static vowels, static vowels with initial coronal or palatal fricatives and vowels embedded in the nonce word [ʃpVp]. We show front and high vowels entail earlier onset of GG activation and that vowels preceded by coronal fricatives are associated with significantly earlier muscle activation than vowels preceded by palatal fricatives. We also note timing differences between GG regions. Thus, for these stimuli, posterior GG EMG activation encompasses the duration of anterior GG EMG activation. These findings underscore the dynamic nature of lingual movement wherein regional tongue muscle activities exhibit predictable differences determined in large part by phonetic context.

8:00

3aSC2. On transition between voice registers: Data from high-speed laryngeal videoendoscopy. Gang Chen, Soo-Jin Park (Dept. of Elec. Eng., Univ. of California, Los Angeles, 63-134 Engr IV, Los Angeles, CA 90095-1594, gangchen@ee.ucla.edu), Jody Kreiman (Head & Neck Surgery, Univ. of California, Los Angeles, Los Angeles, CA), and Abeer Alwan (Elec. Eng., Univ. of California, Los Angeles, Los Angeles, CA)

How specific aspects of vocal fold vibration alter voice register has long been a subject of interest. Transitions between vocal registers are often studied using dynamic vocal fold models and electroglottographic signals. Although laryngeal high-speed videoendoscopy has also been applied to study steady-state voice registers, there has been little such investigation on the transitions between registers. In this study, we examined voice register transitions using phonations in which vocal qualities varied continuously in fundamental frequency (F0), loudness, or voice quality (from breathy to pressed). Glottal area measures {open quotient and alternating-current to open quotient ratios [Chen *et al*., JASA **133**, 1656–1666 (2013)]} and acoustic measures (F0, energy, and H1-H2) were studied using simultaneously collected laryngeal high-speed videoendoscopy and audio recordings from 15 subjects. The video recordings were collected transorally at 10 000 frames per second. Glottal area waveforms were extracted using GlotAnTools (Version 5) [Erlangen, Germany], and acoustic measures were gathered using VoiceSauce software and analysis-by-synthesis. Glottal area measures were compared to acoustic measures and their relationships to voice register transitions were explored.

3aSC3. Duration and rise-span on rising pitch in conversational American English: Rises are never both long and large. Joseph Tyler (Dept. of English Lit. and Linguist, Qatar Univ., Doha P.O. Box 2713, Qatar, josephctyler@gmail.com)

Studies on variation in rising pitch in dialects of English have analyzed correlations of rise-start pitch, rise-end pitch, rise-span, rise-onset position (early vs. late) and pitch dynamism for how they correlate with contextual factors like speaker gender, utterance type (question vs. statement), and turn position (turn-medial vs. turn-final) [e.g., Fletcher and Harrington (2001); Ritchart and Arvaniti (2013); Warren (2005)]. Using data from the Santa Barbara Corpus of Spoken American English, this study adds the novel measure of rise duration. Modeling the data with a linear mixed model with random effects for speaker and conversation (using the lmer function in R), neither gender, utterance type, nor turn position predict rise duration. Nevertheless, rise duration does pattern with the size of the rise (rise-span): many rises are both short and small, some are short and large or long and small, but none are long and large. This suggests a phonetic restriction on the distribution of rises, with speakers avoiding the simultaneous extremes of both duration and span. With the same model testing gender, utterance type and turn position as predictors of rise-span, results show women produce larger rises on questions than statements, while men's rises show no difference between questions and statements.

3aSC4. Intrinsic fundamental frequency (F0) in American English vowels is affected by regional variation. Robert A. Fox and Ewa Jacewicz (SPA Labs, Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu)

There is a relationship between F0 values and vowel height: High vowels have higher F0 than low vowels. A long-standing debate has centered on whether this intrinsic F0 (IF0) pattern is an automatic consequence of vowel articulation or whether it represents deliberate effort to enhance vowel contrast. We provide new data from regional variation suggesting that IF0 is partly controlled by the speaker. F0 was analyzed in high, mid, and low vowels, stressed and unstressed, in 36 females representing dialects spoken in Ohio, Wisconsin, and North Carolina. A robust finding was that dialects differ in their use of IF0 in stressed vowels: IF0 differences between high and low vowels in North Carolina were significantly smaller than those in Ohio and Wisconsin for measurements at vowel onset, offset, peak F0, and overall F0. High vowels had higher IF0 than low vowels in all dialects, confirming the universal aspect of IF0. However, the magnitude of this difference was dialect-specific, which suggests a relative independence of IF0 from vowel height. The lack of correspondence between dialect-specific formant values and dialect-specific IF0 indicates that the quality of vowels and their IF0 value are two features that need to be learned separately.

3aSC5. The phonetic status of [æ] in Twi: A reanalysis. Charlotte F. Lomotey (Lit. and Lang., Texas A&M University-Commerce, 1818D Hunt st., Commerce, TX 75428, cefolatey@yahoo.com)

In Akan, [æ] is regarded as an allophone of /a/ and described as "a quality that ranges from a front vowel quality close to [ɛ] in the Asante [Twi] dialect, to a more central quality in the Fante sub-dialects in which it occurs" (Dolphyne, 1988, p. 6–7). In fact, the existence of [æ] appears controversial because while it is believed to be an allophone of /a/ in the Twi dialect (Boadi, 1991; Dolphyne, 2006; O'Keefe, 2003); and in some sub-dialects of Fante (Abakah, 1978; Boadi, 1991; Dolphyne, 2006), recent studies show that [æ] is a phoneme in Fante (Abakah, 2002; Lomotey, 2008). The present study extends the results of Lomotey (2008), in which she suggests that [æ] may not be an allophone of /a/ in Twi, but another realization of [ɪ] or [e] ($p>0.05$). Results of acoustic analysis (duration, F1, F2) of words with [ɪ], [e] and [æ] in Twi showed similarity with Lomotey (2008). Based on the results, I argue that [æ] may be undergoing or has undergone vowel raising or merger (if it ever exists/existed). I also argue that if it exists in Twi, then it might be represented with a different vowel symbol, and not [æ].

3aSC6. Variability in vowel production: Exploring interactions among frequency, neighborhood density, predictability, and mention. Rachel S. Burdin, Rory Turnbull, and Cynthia G. Clopper (Dept. of Linguist, The Ohio State Univ., 1712 Neil Ave., 222 Oxely Hall, Columbus, OH 43210, burdin@ling.osu.edu)

Multiple factors are known to affect vowel production, including word frequency, neighborhood density, close predictability, and mention. In this study, we explored interactions between the effects of all four of these factors on vowel duration and dispersion in a fully crossed within-subjects design. Participants read a series of short stories that contained target words varying in frequency, neighborhood density, predictability, and mention. Vowel duration and dispersion from the center of vowel space were measured. Results from linear mixed effect modeling revealed the expected effect of neighborhood density on duration: vowels in words with higher neighborhood density were longer. Both frequency and mention had significant expected effects on vowel duration and dispersion: vowels in more frequent words and second mention words were shorter and less peripheral. An interaction between frequency and mention on dispersion was also observed, such that low frequency words underwent second mention reduction to a greater degree than high frequency words. No effects of predictability were observed. These results are only partially consistent with reported findings elsewhere in the literature, suggesting that these effects are fragile and may be substantially affected by methodological choices, including the target words, task, and statistical models.

3aSC7. The implementation of voicing in obstruents in American English connected speech. Lisa Davidson (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, lisa.davidson@nyu.edu)

Descriptions of obstruent voicing in English phonetics textbooks observe that [+ voice] obstruents are likely only fully voiced between other voiced sounds, but have less voicing in word-initial and word-final position [e.g., Cruttenden (2008); Docherty (1992) for experimental data for British English]. This study examines these claims for American English using a corpus of 27 speakers reading 3–5 short stories. Voiced stops and fricatives were coded for preceding and following segments, stress of the previous and following syllables, position within the word (initial, medial, and final), and phrasal position of the word. The calculation of unvoiced frames from the PRAAT voice report was used to obtain the proportion of voicing during each obstruent closure. Results indicate the influence of several phonotactic and prosodic factors. (1) The segment preceding the obstruent influences the amount of voicing more than the following segment does; complete devoicing and shorter durations of partial voicing are most often conditioned by preceding pauses or voiceless sounds. (2) Among following sounds, nasals condition the most complete voicing. (3) An obstruent's position within a word has less effect than its phrasal position. (3) Stress primarily affects stops: a preceding stressed syllable conditions significantly more voicing than a preceding unstressed syllable.

3aSC8. Nuclear pitch accent in American English dialects: Truncation and compression. Ewa Jacewicz and Robert A. Fox (SPA Labs, Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210-1002, jacewicz.1@osu.edu)

Dialects of the same language can differ in their use of prosody. We explored realization of nuclear pitch accent in female speech in three regional varieties of American English: Ohio, Wisconsin, and North Carolina. We found that both OH and WI speakers truncate the F0 contours while NC speakers compress them in identical environments. Our second question pertained to the effects of obstruent voicing in syllable coda on the F0 of the preceding nuclear vowel. We admitted three possibilities for the effects of a voiceless coda: shortened vowel duration may (1) "clip" the F0 contour and reduce the dialectal differences; (2) maintain the dialectal differences by preserving the F0 contour shapes found before a voiced coda; or (3) some dialects may "clip" and some may preserve the F0 contour. The results supported the third option. Ohio and Wisconsin speakers "clipped" the F0 contour so that F0 terminal values before a voiceless coda were higher than before a voiced coda. However, North Carolina speakers not only preserved the contours but their F0 terminal values before a voiceless coda were substantially lower than before a voiced coda. The effects of coda voicing on F0 fall were found to augment the dialectal differences.

3aSC9. Ixpantepec Nieves mixtec question intonation. Younah Chung and Amanda Ritchart (Linguist Dept., UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0108, yachung@ucsd.edu)

Ixpantepec Nieves Mixtec (INM) is posited to have a three-way lexical tone system and lexical stress. However, no known work has investigated whether INM makes use of F0 intonationally. Our study addresses this gap and seeks to contribute to the typology of tone-intonation interactions. We recorded both polar and wh-questions. For polar questions, we analyzed the F0 from one female speaker. The speaker produced question-answer (Q&A) pairs, where the object of transitive verb phrases appeared in both the interrogative and declarative sentences, and differed in its lexical tone. The F0 of every vowel was measured to determine if lexical tones vary depending on whether they occur in an interrogative or declarative sentence of the Q&A pair. Pitch raising is the only significant difference ($p < 0.01$) between the Q&A pair; both high and low lexical tones have higher F0 in the interrogative compared to the declarative. For wh-questions, a qualitative analysis was done using interview data from a male and female speakers. A quantitative analysis is also underway. The Q&A pairs had similar pitch contours and absolute F0 values; thus, no wh-question intonation or tone-intonation interaction was found. Implications for intonational typology in languages with lexical tones will be discussed.

3aSC10. Japanese children's suprasegmental imitation capability of English rhythm. Hiromi Kawai (Ctr. for Teaching English to Children, Kanda Univ. of Int.. Studies, 1-4-1 Wakaba, Mihama-ku, Chiba City 2610014, Japan, kawai-h@kanda.kuis.ac.jp)

This study reports on an acoustic analysis of L2 Japanese child learners' English rhythmic imitation capabilities. The participants for this study were Japanese fifth and sixth graders, all of whom had been taking English language classes since their first year of elementary school. Rhythmic chants were employed as a means to elicit oral production owing to their frequent use in the EFL classroom. Nine English sentences (including two to six stressed syllables, four downbeat sentences and five upbeat sentences) were employed in the collection of a rebus rhyme and in consideration of word familiarity. In the first of two parts of the experiment, the participants shadowed back the test sentences following the experimenter's model pronunciation. In the second part, the experimenter showed picture cards related to the test sentences during the participants' imitation of the experimenter's utterance. During both tests, the experimenter tapped rhythm in time to each participant's utterance of stressed syllables instead of using automatic rhythmic machine. Acoustic analyses and Cluster analysis were used to discriminate between the native-like rhythmic and non-native groups, and profile the 5th and 6th grade groups. Overall, the findings have clear implications for the teaching of English rhythm to moraic language speaking children.

3aSC11. An acoustic analysis of advanced tongue root harmony in Karajá. Sean A. Fulop and Ron Warren (Linguist, California State Univ. Fresno, 5245 N Backer Ave., PB92, Fresno, CA 93740-8001, sfulop@csufresno.edu)

Recent phonological analysis of Karajá, a Macro-Jê language of Brazil, claimed that the language's vowel system evinces advanced tongue root (ATR) harmony [Ribeiro, "ATR harmony and palatalization in Karajá," Santa Barbara Pap. Linguist. **10** (2000)]. Despite this, the phonetic facts about these vowels have never been published, which has left the only claim that ATR operates in any American language subject to controversy. The use of tongue root advancement can never be completely established without articulatory investigation (e.g., MRI scan), but we here provide an acoustic analysis of two native Karajá speakers, which examines the correlates of [ATR/RTR] in four pairs of vowels. An ATR vowel involves expansion of the pharyngeal cavity by moving the base of the tongue forward and/or lowering the larynx during vowel production, and being generally involved in vowel harmony, contrasts with a retracted [RTR] version of the vowel. Acoustic correlates of tongue root advancement generally include a lowering of the frequency of F1 as the pharyngeal cavity expands, together with changes in spectral timbre as measured by the relative formant amplitudes. Particularly, the amplitude of F1 is frequently greater in [ATR] vowels when compared with [RTR] vowels [Fulop *et al.*, "An acoustic analysis of the tongue root contrast in Degema vowels," Phonetica **55**, 80–98 (1998)], and this will be used to illuminate the phonological claims.

3aSC12. Phonetic accommodation in Russian-Estonian bilingual speech. Cameron Rule (Dept. of Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109, rule@umich.edu)

The influence of L1 phonological knowledge on L2 bilingual speech has long been a fruitful topic of speech perception and production research (Piske *et al.*, J. Phonetics **29**, 191–215). Recent research examining phonetic accommodation in bilingual speech has demonstrated that L1 and L2 phonologies interact in a bidirectional, dynamic manner (Fowler *et al.*, J. Phonetics **36**, 649–663). My study builds on these results by examining the acoustic outcomes of phonetic accommodation in vowels produced by bilingual Russian(L1)-Estonian(L2) speakers. Specifically, I aim to test the hypotheses that L2 knowledge can significantly affect robust aspects of L1 speech, i.e., vowel quality, and that there is a correlation between the extent of L2 exposure and the degree of phonetic accommodation. The preliminary results of acoustic analysis suggest that bilingual speakers produce L1 vowels with formant values that are intermediate from both L1 and L2 monolingual controls, indicating that L2 exposure influences significant aspects of L1 speech. Presently, I am conducting analyses which compare individual speaker differences in the extent of phonetic convergence to the amount of their L2 use and exposure. My overarching objective is to present novel empirical evidence that expands our current understanding of phonetic accommodation in bilingual speech production.

3aSC13. Intelligibility in dysphonic speech: Landmark-based measures. Suzanne Boyce, Keiko Ishikawa (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, 5371 Farmridge Way, Mason, OH 45040, ishikak@mail.uc.edu), and Marisha Speights (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Speakers with voice disorders frequently report reduced intelligibility in ordinary communication situations. This effect is typically attributed to reduced loudness; however, other source/vocal tract interactions may be at work. The acoustic landmark theory of speech perception postulates that specific acoustic events, called "landmarks," contain particularly salient information about acoustic cues used by listeners. The current study examined acoustic profiles of dysphonic speech with the publically available landmark-based automatic speech analysis software, SpeechMark^{TM}. In this study, we focused on burst landmarks, which aim to identify onsets and offsets of affricate/stop bursts. The study tested two hypotheses: (1) normal and dysphonic speech samples differ in the number of burst landmarks because laryngeal pathology affects the consistency of airflow control, and (2) the number of burst landmarks will correlate with cepstral peak prominence values, which have been shown to correlate well with perceptual judgment of dysphonia severity. Speech samples of 36 normal and 33 dysphonic speakers from KAY Elemetrics database of Disordered Voice were subjected to the analysis. Results will be discussed in the context of clinical assessment of intelligibility for dysphonic voices.

3aSC14. The merged vowel of PIN and PEN as realized in Bakersfield, California. Ron Warren and Sean A. Fulop (Linguist, California State Univ. Fresno, 5245 N. Backer Ave., Linguist PB92, Fresno, CA 93740-8001, warlockwarren@gmail.com)

One consistently identified quality of Southern American English (SAE) is the PIN/PEN merger, wherein the vowels /ɪ/ and /ɛ/ become indistinguishable in nasalized contexts. Existing studies have shown that the merged vowel may either occupy a space between the vowel spaces of /ɪ/ and /ɛ/ [Bigham, MA thesis, UT Austin (2005)], or merge completely into /ɪ/ [Koops *et al.*, "The effect of perceived speaker age on the perception of PIN and PEN vowels in Houston, Texas," U Penn Working Pap. Linguist. **14**(2) (2008)]. Due to the dustbowl migration, areas of the southern central valley of California have a complete PIN/PEN merger similar to SAE. This study examines the acoustic properties of the merged vowel spoken by eight speakers from Bakersfield, California, ages 25 through 65. Findings indicate that the merged vowel does not occupy a compromise position, nor a merged /ɪ/ position in any but the oldest subject. Instead, the merged vowel presents significantly lower first formant values than /ɪ/, suggesting that as the California vowel shift has lowered the front lax vowels, this merged allophone has remained stationary. Further, the youngest female subjects presented an inconsistent vowel merger, suggesting a potential demerger of the type documented in southern urban centers. This raises questions about the nature of the alleged merger, if perceptible acoustic differences remain which allow for demerger.

3aSC15. Linearly diverging vocal folds and the Coanda effect. Jesse Haas, Xavier Pelorson (Parole et Cognition, INPG, Universite Stendhal, CNRS, Domaine Universitaire, BP46, Saint Martin d'Heres 38402, France, jesse.haas@gipsa-lab.grenoble-inp.fr), and Avraham Hirschberg (Technische Universiteit Eindhoven, Zwartenberg, Netherlands)

We would like to verify whether the Coanda effect has a significant impact when incorporated into theoretical vocal fold models that assume a piecewise linear shape of the vocal fold walls, as many do. We model the intraglottal flow with the equations of Thwaites. Thwaites boundary layer theory gives simple criteria for the glottal jet separation point if the vocal folds diverge linearly, and even validates well-known empirical observations for the jet width at flow separation. We test this criteria against flow experiments with rigid vocal fold replicas. The experiments involve symmetric and asymmetric vocal fold configurations, as well as steady and unsteady flow. We then validate the significance of the predicted Coanda effect on several numerical models of human vocal folds. We test the significance of the effect both on mechanically symmetric vocal fold models and on ones with mechanical asymmetries. We find limited effects on symmetric vocal folds and varying degrees of impact on asymmetric vocal folds.

3aSC16. Realization of stress in theatrical speech: The example of Composed Theater. Kostis Dimos (Phonetic Lab., Dept. of General Linguist, Univ. of Zurich, Plattenstrasse 54, Zurich CH-8032, Switzerland, kostis.dimos@uzh.ch), Leopold Dick (Bern Univ. of the Arts, Bern, Switzerland), and Volker Dellwo (Phonetic Lab., Dept. of General Linguist, Univ. of Zurich, Zurich, Switzerland)

In this study, we investigated syllabic stress in theatrical speech containing emphatically stressed vowels. Typical acoustic correlates of stress are intensity, pitch, and duration. The aim of this experiment was to measure whether artistic emphatic stress is realized differently in terms of these correlates compared to stress in normal speech. In an experiment, one professional performer of Composed Theater, an avant-garde type of experimental theater, was recorded both in natural and theatrical French speech. The results showed an increase in mean intensity, pitch, and mean duration in highly stressed vowels. The difference between stressed and unstressed syllables was higher in theatrical, than in normal speech. We argue that duration, pitch and intensity were used to different degrees to indicate stress in the artistic speech sample. Our study provided a preliminary comparison of

speech characteristics between artistic and normal speech focusing on stress realization techniques in artistic speech performance. Additional professional speakers' recordings are currently being analyzed in terms of prosodic characteristics in avant-garde theater performance.

3aSC17. Contextual landmark analysis of speech from typically and atypically developing children. Chelsea Levy, Allison Mann, Jess Kenney, Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Res. Lab. of Electronics, MIT, 50 Vassar St., Rm. 36-581, Cambridge, MA 02139, jyechoi@mit.edu)

Analysis of acoustic landmarks (abrupt changes in the speech signal spectrum, Stevens 2002) has been carried out for speech from two typically and six atypically developing children. The atypically developing children include two with specific language impairment (SLI), and four diagnosed with autism spectral disorder, and two with higher and two with lower language functions (ASDH and ASDL, respectively). Recordings of non-word repetition sessions (CNREP) for each child were hand annotated with words, phones, and landmarks (42 tokens per child). Decision tree analysis was used to examine the effects of factors such as landmark type, preceding and following phone type, position in the syllable (onset, nucleus, ambisyllabic, and coda), and syllabic stress (stressed, full, and reduced) on landmark modification patterns. Results indicate that, compared with typically developing children, atypically developing children show different landmark modifications, and that these different modification patterns can be characterized by the systematic effects of contextual factors.

3aSC18. Measures of spectral tilt in Shanghainese stops and glottal fricatives. Laura L. Koenig (Haskins Labs and Long Island Univ., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu) and Lu-Feng Shi (Long Island Univ., Brooklyn, NY)

Shanghainese differs from other major Chinese dialects in having a three-way contrast among stop consonants. Although phonological descriptions often label these sounds as voiceless aspirated, voiceless unaspirated, and voiced, in contemporary Shanghainese the "voiced" category is not typically produced with closure voicing in initial position. Rather, the vowel onset appears to be characterized by breathy voicing, as demonstrated by acoustic, aerodynamic, and transillumination data. Past studies have been limited to few speakers. This study presents preliminary data of a large-scale study of the Shanghainese stop system as well as the voicing contrast in glottal fricatives, another unusual feature of the language. Data have been collected from 20 male and female native speakers. Measures are made of voice onset time (for the stops), and three measures of spectral tilt (for stops and glottal fricatives): The relative amplitudes of (a) the first two harmonics (H1-H2); (b) H1 and the first formant (H1-A1); and (c) H1 and the third formant (H1-A3). Spectral tilt measures are taken at phonation onset and 50 ms into the vowel. Preliminary analyses suggest that all three spectral measures may lend insight into the nature of the source distinction among the stops as well as the glottal fricatives.

3aSC19. Assessing the effects of cognitive decline on the speech rate of demented and healthy elderly speakers. Maria Heffinger, Gina DAmico, Linda Carozza (Commun. Sci. and Disord., St. John, Staten Island, NY), Fredericka Bell-Berti, and Pamela C. Krieger (Commun. Sci. and Disord., St. John, 8000 Utopia Parkway, Queens, NY 11439, bellf@stjohns.edu)

According to current research, both age and cognitive decline have an effect on speech rate. While healthy elderly speakers are reported to speak at a slower rate than their younger counterparts, it has also been noted that elderly patients with a cognitive decline speak more slowly than their healthy elderly peers. This study analyzes the speech rates of healthy elderly speakers and compares the results with our previously reported data from speakers diagnosed with dementia. Our healthy elderly participants repeated five-syllable carrier phrases with varying target words. The durations of consonant and vowel segments will be measured using digital spectrograms to determine the extent of final lengthening, compensatory shortening, and overall speech rate (in syllables and second). Results for the two speaker groups will be compared with each other and also with results previously reported in the literature.

3aSC20. An acoustic analysis of lexical stress in non-words in Uyghur. Mahire Yakup (Univ. of Kansas, 2211 Willow Crk, Lawrence, KS 66049, mylyakup@ku.edu)

Previous research examined the stress pattern in Uyghur, a Turkic language, using real words and found that Uyghur used the duration as a stress cue but not F0, and the intensity was moderated by syllable types. In order to avoid vowel lengthening in the real words, the non-words (DOSdos vs. dosDOS or DOdo vs. doDO, capitalized as stressed) were used, in which the syllable types (CVC syllable and CV syllable) and vowels (a, u, i, o, ø, y) were controlled. In the production experiments, speakers were clearly instructed that the capitalized ones were stressed and lower-cased as unstressed syllables. All target words were embedded in the same carrier sentence (e.g. 'I will say X now.'). In the production of ten female native Uyghur speakers, average fundamental frequency, duration, average intensity, and first and second formant frequencies for vowels were collected in the accented and unaccented syllables. The results showed that there were significant differences in duration and intensity between stressed and unstressed syllables. The fundamental frequency differences were associated with word final positions rather than stress positions. Vowels are centralized in the unstressed position. The present acoustic data suggest that native Uyghur speakers use of duration and intensity rather than F0.

3aSC21. Effects of predictability on vowel reduction. James D. Whang (Linguist, New York Univ., 10 Washington Pl, New York, NY 10003, james.whang@nyu.edu)

High vowel devoicing in Japanese, where unaccented /i, u/ in a C_1VC_2 sequence reduce when both C_1 and C_2 are voiceless, has been studied extensively, but whether the target vowel is truly devoiced (oral gesture is maintained) or deleted completely (oral gesture and voicing are both lost) is still debated. This study examines the effects of vowel predictability on the degree of vowel reduction. Native Japanese speakers ($N = 8$) were recorded in a sound-proof booth reading sentences containing lexical stimuli. C_1 of the stimuli were [k, ʃ], after which either high vowel can occur, and [ϕ, s, ç], after which only one of the two is possible. Half of the stimuli contained a devoicing environment with a voiceless C_2. Center of gravity, the amplitude weighted mean of frequencies present in a signal, was measured for the first half (COG1) and the second half (COG2) of C_1. Results show that COG2 is significantly lower than COG1 for all consonants when a full vowel follows, as well as for [k, ʃ] in devoicing stimuli. In contrast, COG remained stable for [ϕ, s, ç] in devoicing environments, suggesting a complete lack of vowel gestures. Predictable vowels, therefore, seem to delete, while unpredictable vowels devoice.

3aSC22. Lung volume initiation levels and selected acoustic measures of English consonants. Peter J. Watson and Yu-Wen Chen (Speech-Language-Hearing Sci., Univ. of Minnesota - Twin Cities, 164 Pillsbury Dr., Shevlin 115, Minneapolis, MN 55455, pjwatson@umn.edu)

Watson, Ciccia, and Weismer (2003) described the relationship of initiating speech at different lung volumes to selected acoustic variables related to vowel production. It was found that some variables, such as dB SPL were related to a 'direct' mechanical interaction with the breathing system. They also found that a variable, such as vowel-space, was reduced at low-lung volume initiation levels suggesting an indirect link between one subsystem, breathing, and another articulation. Using a similar procedure to Watson *et al.* (2003), we studied selected acoustic variables of English fricatives and stops. Participants read aloud a carrier phrase with a 2 syllable V—'CV embedded within it. Participants were trained to initiate speech at 3 different lung volume levels: normal, low, and high. Data will be discussed in relation to those acoustic variables that have a more direct interaction with the breathing system, e.g., db SPL, and those with a less direct relationship with breathing, e.g., relative duration of voice-onset-time between cognate pairs of voiced and voiceless stop consonants, and the difference of first moment measures between sibilants.

3aSC23. Effects of speaking mode (clear, habitual, slow speech) on vowels of individuals with Parkinson's disease. Rebekah A. Buccheri (Dept. of Speech-Lang. Pathology/Audiol., Molloy College, 1000 Hempstead Ave., W104, Rockville Ctr., NY 11571, rbuccheri@molloy.edu), Douglas H. Whalen, Winifred Strange (Speech-Language-Hearing Sci., The Graduate Ctr. (CUNY), New York, NY), Nancy S. McGarr (Dept. of Speech-Lang. Pathology/Audiol., Molloy College, Rockville Ctr., NY), and Lawrence J. Raphael (Commun. Sci. and Disord., Adelphi Univ., Garden City, NY)

This study examined the effects of three different speaking modes (clear, habitual, and slow speech) on speech production of individuals with and without Parkinson's disease. Twenty-one speakers (13 with Parkinson's, 8 Controls) read the Farm passage in habitual, clear, and slow speech modes. Acoustic analysis involving the assessment of the first and second formant frequencies was performed using: vowel space areas, vowel dispersions, /i-ɑ/ distances measures for both tense and lax vowels produced in each of the speaking conditions. Results revealed that for both groups, the vowel space areas were larger in the clear and slow conditions compared to habitual, with no difference between clear and slow for tense vowels. However, there was no significant difference across any of the speaking conditions for lax vowels. There was a significant difference in vowel dispersion measures, between the habitual and clear speech conditions and also the slow and habitual conditions. But there was no significant difference between the clear and slow conditions for vowel dispersions. With respect to /i-ɑ/ distance, there was a significant difference between the habitual condition and clear speech. Implications of these results will be presented for both Parkinson's disease subjects and normal controls.

3aSC24. Segmental and prosodic effects on intervocalic voiceless stop reduction in connected speech. Dominique A. Bouavichith (Dept. of Linguist, New York Univ., 33 Washington Sq W, 1115, New York, NY 10011, dab491@nyu.edu)

Descriptions of English and other languages have claimed that voiced and voiceless intervocalic stops are often lenited to fricatives and approximants in connected speech. Few acoustic analyses of factors that affect this reduction have been reported for American English [cf. Lavoie (2001), Tucker and Warner (2011)]. In this analysis, intervocalic voiceless stops produced in bisyllabic words during story reading are examined (participants N = 19). The first result shows that speakers never lenite voiceless stops to approximants, except when /t/ is produced as the approximant implementation of a flap. This shows that voiced and voiceless stop reduction behave differently. Second, stress and vowel reduction play an integral role: 33% of stops are produced as fricatives when stress is on the preceding syllable (e.g., "taco"); 4% when stress is on the following syllable (e.g., "account"). The rate of reduction is significantly lower when stops are surrounded by two full vowels and higher when they are followed by schwa. Third, fricative reduction is most common for /k/, since full closures may be most difficult in the velar region. This study will be compared to Bouavichith and Davidson's (2013) examination of voiced stops to provide a fuller picture of intervocalic stop reduction in American English.

3aSC25. Temporal alignment between head gesture and prosodic prominence in naturally occurring conversation: An electromagnetic articulometry study. Dolly Goldenberg (Linguist, Yale Univ., 192 Foster St., Apt. 1, New Haven, CT 06511, dolly.goldenberg@yale.edu), Mark Tiede, Douglas N. Honorof (Haskins Labs., New Haven, CT), and Christine Mooshammer (Institut für deutsche Sprache u. Linguistik, Berlin, Germany)

Studies of the relationship between speech events and gesticulation have suggested that the peak of the prosodic pitch accent serves as a target with which body gestures may be coordinated (Roth, 2002; Loehr, 2004). While previous work has relied on controlled speech elicitation generally restricted to nonrepresentational extension/retraction (Leonard and Cummins, 2011) or iconic (Kelly *et al*., 2008) gestures, here we examine the kinematics of the speech articulators and associated head movements from pairs of individuals engaged in spontaneous conversation. Age and gender matched native speakers of American English seated 2 m apart were recorded using two electromagnetic articulometer (EMA) devices (Tiede and Mooshammer, 2013). Head movements were characterized by the centroid of reference sensors placed on the left and right mastoid processes and the upper incisors. Pitch accents were coded following the ToBI implementation of Pierrehumbert's intonational framework following Beckman and Elam (1997). Preliminary findings show that the apex (point of maximum excursion) of head movements within an IP in general precedes the peak of the associated pitch accent, and is consistently aligned with co-occurring articulatory events within the syllable. [Work supported by NIH NIDCD-DC-012350.]

3aSC26. Production and perception of the English sibilants /s/ and /ʃ/ in persons with Parkinson's disease. Yu-Wen Chen and Peter J. Watson (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Shevlin Hall 115, Minneapolis, MN 55455, chen1887@umn.edu)

Parkinson's disease (PD) presents with both sensory and motor deficits including speech, characterized by soft voice, monotonicity, and imprecise articulation. Both audition and somatosense are believed important for accurate speech production. Individual studies have identified auditory impairment and somatosensory impairment of the orofacial structures and the larynx. No studies have examined the relationship between imprecise articulation and sensory impairment. This research aims at examining sensory deficits and the relationship between sensory deficits and imprecise articulation in persons with PD. The production of /s/ and /ʃ/, auditory discrimination and identification in spectral shape, and somatosensory acuity on the tongue tip are examined. In the production task, participants read monosyllabic words beginning with /s/ and /ʃ/ embedded in a carrier sentence; measures of spectral shapes are made. Auditory tasks comprise an acuity task testing for participants' auditory acuity in perceiving spectral shapes and a /s/-/ʃ/ categorization task testing for their categorical boundaries. The somatosensory acuity task tests for participants' somatosensory acuity in spatial orientation on the tongue tip. Relationships between production of /s/ and /ʃ/ and sensory performance are tested in correlational tests.

3aSC27. Relationship between the first two formant frequencies and tongue positional changes in production of /aɪ/. Jimin Lee (Commun. Sci. and Disord., Penn State Univ., 404A Ford Bldg., University Park, PA 16802, jxl91@psu.edu)

The first two formant frequencies (F1, F2) of vowels are often interpreted in terms of their relationship to tongue height and advancement, respectively. To test this interpretation, the current study examines the relationship between F1/F2 trajectories and tongue positional changes in production of the diphthong /aɪ/ by utilizing electromagnetic articulography. Ten healthy female speakers participated in the current study. Electromagnetic articulography (AG-200) and a synchronized audio recording system were utilized to obtain synchronized kinematic and acoustic data. Each speaker produced three repetitions of the word "hide" in the carrier phrase "I say a_ again." F1 and F2 were traced along the entire vocalic nucleus of the target vowel /aɪ/. Over the same time interval, x and y coordinate values from the tongue sensor (positioned approximately 25 mm away from tongue apex) was recorded. Correlational analysis (r-values) showed that, overall, F1—tongue y position and F2—tongue x position pairs have strong relationships; however, a strong relationship was observed in F1-x and F2-y pairs as well. Results will be discussed in terms of amount of variance of formant frequencies explained by tongue xy positional changes and issues of interpretation of formant frequencies.

3aSC28. Vowel and consonant effects on subglottal pressure. Didier Demolin, Silvain Gerbers (Gipsa-lab, Université Stendhal, 1180 Ave. Centrale, Grenoble Grenoble cedex9, France, didier.demolin@gipsa-lab.grenoble-inp.fr), and Sergio Hassid (Gipsa-lab, Université Stendhal, Brussels, Belgium)

The respiratory system is generally regarded as producing voluntary variations in intensity and perhaps in pitch, but not producing voluntary increases in pressure for particular sounds. All the changes related to individual segments, such as the drop in subglottal pressure that occurs after [h] or the increase in pressure during the [k] closure are considered to be aspects of tract aerodynamics, and not under voluntary control. They can be

ascribed to variations in the resistance provided by the vocal folds to the outgoing air (the glottal impedance) or to variations in the stiffness of the vocal tract walls. This paper examines variations accompanying different vowels and consonants, and concludes that it is not only the Koreans who use greater respiratory effort to distinguish some sounds. The principal finding of this study was that there was a considerable difference in the subglottal pressure associated with the different vowels. Across all other variables, the difference in subglottal pressure between /a/ and /i/ is 1.65 cm hPa and between /a/ and /u/ is 1.76 hPa. Both these differences are highly significant ($p < 0.0001$) in a one way ANOVA. The difference between /i/ and /u/ is 0.11 hPa and is not significant ($p = 0.303$).

3aSC29. Phonation and tone in conversational Beijing Mandarin. Patrick R. Callier (Dept. of Linguist, Stanford Univ., Bldg. 460, Stanford, CA 94305, pcallier@stanford.edu)

Previous acoustic analyses of phonation in Beijing Mandarin and closely related varieties have shown that phonatory variability in Mandarin is conditioned by lexical tone and prosody. Tone 3 (low) and tone 4 (high fall), as-well as syllables in domain-final position, have been reported to have more negative spectral tilt (particularly lower H1-H2), indicating creaky phonation. This study examines phonation in a corpus of 6752 vowels from conversational interview speech with 16 university students in the Beijing area, using four measures of spectral tilt: H1-H2, H1-A1, H1-A2, and H1-A3. The results, which diverge from findings based on laboratory speech, are possibly the first to report on spectral tilt for tone 5 (neutral tone). In mixed-effects regression models, tone 2 (high rising) has lower H1-A1, tone 3 and tone 4 have lower H1-A1 and H1-A2, and tone 5 has lower values of H1-H2 and H1-A1. Prosodic position interacts with tone. IP-final syllables for each tone exhibit higher spectral tilt on some measures than non-final syllables, though main effects for prosodic position give lower values in IP-final position for H1-A1, H1-A2, and H1-A3. These results encourage more attention to the interaction of prosodic and tonal factors in naturalistic speaking contexts when studying phonation.

3aSC30. Acoustic measures of falsetto voice. Patricia Keating (Linguist, UCLA, Linguist, Los Angeles, CA 90095-1543, keating@humnet.ucla.edu)

Falsetto (or loft) voice is known to be generally characterized by a high fundamental frequency, a spectrum with relatively few harmonics, and a relatively strong fundamental harmonic. The present study tests the hypotheses that falsetto voice differs from modal voice not only in the acoustic measure H1*-H2* (the amplitude difference between the first and second harmonics, corrected for formant frequencies and bandwidths), but also (1) in H2*-H4* (the amplitude difference between the second and fourth harmonics, corrected for formant frequencies and bandwidths), and (2) in spectral noise as measured by harmonics-to-noise measures. These and other acoustic measures will be obtained for selected pairs of vowels using VoiceSauce, the UCLA program for voice analysis. The test corpus comprises pairs of sentences taken from two readings of the story "Little Red Riding Hood" by 17 speakers of American English (8 women, 9 men). Each speaker read the story first in a neutral voice, and then using character voices for the dialogs, which for these 17 speakers were often in falsetto. The goal of the study is a reliable acoustic measure of falsetto voice.

3aSC31. Phonetic marking of stance in a collaborative-task spontaneous-speech corpus. Valerie Freeman, Gina A. Levow, and Richard Wright (Linguist, Univ. of Washington, Box 352425, Seattle, WA 98195-2425, rawright@uw.edu)

While stance-taking has been examined qualitatively within conversation and discourse analysis and modeled using text-based approaches in computational linguistics, there has been little quantification of its acoustic-phonetic correlates. One reason for this is the relative sparsity of stance-taking behavior in spontaneous conversations. Another is that stance marking is embedded into a highly variable signal that encodes many other channels of information (prosody, word entropy, audience, etc.). To address these issues, we draw on varying subfields to build a corpus of stance-dense conversation and develop methods for identification and analysis of stance-related cues in the speech signal. In the corpus, dyads are engaged in three collaborative tasks designed to elicit increasing levels of investment. In these imaginary store inventory, survival, and budget-balancing scenarios, participants solve problems, but the conversation is otherwise unscripted. Based on limited previous work (Freeman, under review) and initial findings from our corpus, we predict that stance-marking employs hyperarticulation (or lack of reduction) analogous to topic or contrast focus but where reduction would be expected in the discourse structure. Stance-taking is expected to correlate with slower speaking rates, longer stressed vowels, more expanded vowel spaces, greater pitch excursions, and greater modulation of speech signal intensity.

3aSC32. The prosody-pragmatics interface: An acoustic analysis of contrasting Spanish varieties. Ryan Platz (Spanish, Italian and Portuguese, The Penn State Univ., 150 Dorchester Ln., Bellefonte, PA 16823, ryanplatz@gmail.com)

Investigating correlations between communicative contexts and prosodic elements allows researchers to exploit the (perhaps inherent) connection on the prosody-pragmatics interface. Pitch accent models such as ToBi and RaR are helpful for a descriptive analysis, but categorization of different pitch contours can be subjective and rely on the eye of the investigator. Comparing the prosodic elements that define these models' categorization parameters would provide a better approximation of the intonation patterns across specific languages varieties and pragmatic contexts. This study investigates the prosody-pragmatics interface from an acoustic perspective in an analysis of native speakers of several Spanish varieties. Participants responded to 12 situations in which two pragmatic variables were controlled for (social distance, ranking of imposition). Responses were analyzed with a Praat script to capture pitch range, pitch span and speaking rate. Results do reveal a correlation between dialect and each suprasegmental variables, but an even stronger correlation between speaker and the contextual pragmatic variables was found, irrespective of dialect. These findings provide empirical evidence that prosodic elements of the speech signal are more relevant to the given pragmatic context rather than to speaker dialect traits, suggesting intonation patterns are constructed within a communicative context rather than within the speaker's community.

3aSC33. Using developmental data to explore frequency effects in production. Melissa M. Baese-Berk (Dept. of Linguist, 1290 Univ. of Oregon, 217 Agate Hall,Eugene, OR 97403, mbaesebe@uoregon.edu) and Katherine White (Dept. of Psych., Univ. of Waterloo, Waterloo, ON, Canada)

Low-frequency words are produced with longer duration and more extreme articulation than high-frequency words. It is unclear whether these differences are the product of online processes occurring during production, or instead result from the nature of stored exemplars. One way to disentangle these accounts is to consider changes over development. Online accounts predict that frequency effects arise due to the structure of the lexicon. Therefore, as the lexicon develops, productions should change as a function of individual words' frequencies. In contrast, exemplar accounts hypothesize that these effects occur because listeners are exposed to and store words with these acoustic properties; on these accounts, the effects should remain relatively stable across the lifespan. We asked children to name pictures that differed in frequency in both the child and adult lexicons. The pictures were in one of four categories: high frequency adult/high frequency children, low frequency adult/low frequency children, high frequency adult/low frequency children and low frequency adult/high frequency children. If children's production patterns reflect frequency in the adult lexicon, this would be evidence for an exemplar account of frequency effects; if, instead, children's patterns reflect frequency in the child lexicon, this would suggest that online processes underlie these effects.

Session 3aSP

Signal Processing in Acoustics: Intelligent Feature Selection Methods for Machine Learning Problems in Acoustics

Eric A. Dieckman, Cochair
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Ning Xiang, Cochair
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Invited Papers

9:00

3aSP1. Automated classification of oncoming ground vehicles using acoustic echolocation and supervised machine learning. Eric A. Dieckman (Sonalysts, Inc., 84 Nicoll St, Unit 1, New Haven, CT 06511, eric.dieckman@gmail.com) and Mark K. Hinders (Dept. of Appl. Sci., College of William and Mary, Williamsburg, VA)

In order to perform useful tasks, robots must have the ability to notice, recognize, and respond to objects in their environment. This requires the acquisition and synthesis of information from a variety of sensors. Here we focus on acoustic echolocation measurements of approaching vehicles, where an acoustic parametric array propagates an audible signal to the oncoming target and the reflected backscattered signal is recorded using the Microsoft Kinect microphone array. Although useful information about the target is hidden inside the noisy time domain measurements, the Dynamic Wavelet Fingerprint process (DWFP) is used to create a time-frequency representation of the data. Intelligent feature selection allows the creation of a small-dimensional feature vector that best differentiates between vehicle types for use in statistical pattern classification routines. Using experimentally measured data from real vehicles at 50 m, this process is able to correctly classify vehicles into one of five known classes with 94% accuracy. Fully three-dimensional simulations allow us to study the nonlinear beam propagation and interaction with real-world targets to improve classification results.

9:20

3aSP2. Use of supervised machine learning for real-time classification of underwater targets using Autonomous Underwater Vehicle sampled bistatic acoustic scattered fields. Erin M. Fischell and Henrik Schmidt (Mech. Eng., MIT, 77 Massachusetts Ave., 5-204, Cambridge, MA 02139, emf43@mit.edu)

A method has been developed for the classification of underwater target geometry using bistatic acoustic amplitude data collected by an Autonomous Underwater Vehicle (AUV) as it follows a selected path through the scattered field created by a fixed source insonifying a target. The mobility of an AUV allows it to exploit features of this field in three dimensions. The classification method presented includes offline and onboard processing components, which use a combination of signal processing, vehicle behaviors, and machine learning in the form of Support Vector Machines (SVMs) to extract target geometry from collected acoustic data. The offline training and analysis step creates training and test vector sets in a selected feature space from existing scattered field data and outputs models for target classification, confidence, and feature ranking. Several algorithms are explored for selecting the feature space used by the SVM. The models produced by the offline processing step are used in the real-time classification processing chain onboard an AUV sampling an unclassified target's scattered field. The presented simulation results use scattered fields modeled using OASES-SCATT and demonstrate real-time processing and path planning in the LAMSS MOOS-IvP simulation environment. [Work supported by ONR Code 321 OA and NSF GRFP.]

Contributed Papers

9:40

3aSP3. Learning environmentally dependent feature representations for classification of objects on or buried in the seafloor. Jason E. Summers (Appl. Res. Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com), Timothy C. Havens (Depts. of Elec. and Comput. Eng. and Comput. Sci., Michigan Technolog. Univ., Houghton, MI), and Thomas K. Meyer (Appl. Res. in Acoust. LLC, Culpeper, Virginia)

Classification performance of underwater objects is determined in large part by the acoustic response of the environment. This is a consequence of environmental clutter and of the altered response of objects that occurs through interaction with the environment—particularly for low-frequency scattering from objects on or buried in the seafloor. Therefore, to achieve robust classification, it is necessary that signal-processing algorithms account for the local context of the underwater environment. Theoretical, experimental, and numerical studies have characterized well the influence of the environment on scattering from objects for a variety of burial conditions, object shapes, structures, and materials, and sediment types and topographies. However, the relationship between these results and measurable features useful for classification remains to be fully developed. Here, we describe a machine-learning approach to selecting features for environmentally adaptive classification that seeks to algorithmically select from measured and modeled data those feature representations best suited to object

classification in varying environmental contexts. We follow a data-driven online learning approach that, while leveraging current theories and models describing environment/target interaction, does not impose an artificial limitation on the true diversity of targets and their environments. Preliminary results are presented for measured and modeled data.

9:55–10:15 Break

10:15

3aSP4. A method for deconstructing the relaxation absorption spectrums of gas mixtures. Tingting Liu, Shu Wang, and Ming Zhu (Dept. of Electronics and Information Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei 430074, China, liu.tt199104@gmail.com)

It is acknowledged that the relaxation absorption spectrum curves can be employed to detect gas compositions. However, it still remains a challenge to extract the composition information from the spectrums. In this paper, a method is proposed for deconstructing the relaxation absorption spectrums of a gas mixture. Based on the deconstruction results, a gas relaxation absorption spectrum is constituted by the curve tendency, which is determined by the gas effective specific heat, and the curve position, which is determined by the gas effective relaxation time. And the effective specific heat of a gas mixture is the simple addition of the compositions' independent effect, while the effective relaxation time is the complex interaction effect between different compositions. Consequently, the deconstruction method can deconstruct the relaxation absorption spectrums of a gas mixture into the simple sum of its compositions' spectrums. By the comparison between the theory curves and the experimental data, the validity of the deconstruction method is proved. The method holds great potentials for the obtainment of composition information in gas detection.

10:30

3aSP5. Selected prosody features in an accent learning system. Xiao Perdereau (Burgundy Univ., 9, Av. A. Savary, BP 47870, Dijon, France, xiao.chen-perdereau@u-bourgogne.fr)

Human speech interaction experiments have been structured as temporal and spectral representations of acoustical waveforms to better understand human facilities in handling previously unheard non-native accents of a natural language. We used Mandarin as target language for native and non-native speakers. Identical linguistic materials have been applied to both human and machine learning process. The word sequences have been selected for inference. The outcomes from human speech interactions are analyzed using computer. Speech prosody characteristics have been selected to identify and to classify naturally accented speeches. Correct and incorrect word sequences for each accented speech are discriminated. Starting with a limited stored samples of natural speech data, new signals of accented speech could be detected. Relevant acoustic features extracted from human speech are used in a comparative processing algorithm. Modified speech prosodies are produced artificially by modulating a relatively small number of physical parameters. On one hand, the variabilities regenerated by the dynamic device are fed foreword for extended human accented speech training, on the other hand, recycled acoustic signal acquisitions improve our machine knowledge source through gradual accumulation. This work is part of the development of multimedia tools integrated in an accent learning systems.

10:45

3aSP6. Buzz, squeak, and rattle noise classification by using acoustic-fingerprinting technology. Dae-Hoon Seo, Jung-Woo Choi, and Yang-Hann Kim (Mech. Eng., Korea Adv. Inst. of Sci. and Technology(KAIST), 373-1 Guseong-dong, Yuseong-gu, Daejeon 305-701, South Korea, ihuny@kaist.ac.kr)

The buzz, squeak, and rattle (BSR) noises are three representative types of the automotive interior or exterior noise. Some of BSR noises have very short duration in time, and hence, it is difficult to detect various BSR noises in low SNR situation. However, each BSR noise signal has a unique time-frequency characteristic, depending on the various contacting materials as well as the excitation forces. Therefore, it is necessary to utilize the time-frequency characteristic of the BSR noise to specify the origin of a noise source. In this paper, we propose a novel method and system for identifying BSR noises. For accurate classification of noise sources, a noise-fingerprinting and matching technique based on the pattern classification is devised. The identification test with the real BSR noise data shows that the proposed method can accurately classify the noise source even in the low SNR condition.

11:00

3aSP7. Spectrotemporal Gabor filters for feature detection. Leslie S. Smith and Andrew K. Abel (Computing Sci. and Mathematics, Univ. of Stirling, Stirling, Scotland FK9 4LA, United Kingdom, l.s.smith@cs.stir.ac.uk)

Features (landmarks) in sound are located in time and spectrum. Two dimensional (time x spectrum) Gabor filters can be used to detect useful classes of these. We use a set of logarithmically spaced bandpass filters whose outputs are coded as spikes to perform spectral analysis. These are convolved with Gabor filters to create spectrotemporal feature maps. Using auditory-nerve like spikes to code zero-crossings retains precise timings and amplitude information, and makes convolution computation relatively straightforward. Gabor patches with "horizontal" bars (parallel to time axis) can be used to detect harmonicity, and patches with "vertical" bars (parallel to spectrum axis) can detect envelope modulations. Because the time resolution is maintained in the preprocessing, vertical bars may be close together (e.g., 5 to 10 ms apart), enabling detection of amplitude modulation due to unresolved harmonics. This is useful for both for speech voicing detection, and for animal utterances. Such filters may be localized in spectrum, allowing tracking of voicing energy. Filters with bars at other angles can detect frequency modulation. Using constellations of these features (and others, such as onsets), we can characterize and interpret sound sources.

Session 3aUWa

Underwater Acoustics: Acoustic Signal and Noise Propagation and Scattering

Kathleen E. Wage, Chair
George Mason Univ., 4400 University Dr., Fairfax, VA 22030

Contributed Papers

8:00

3aUWa1. Acoustic scattering from a sand layer and rock substrate with rough interfaces using the finite element method. Anthony L. Bonomo, Marcia J. Isakson, and Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

The finite element method is used to study the acoustic scattering from a layer of sand overlying a rock substrate. All the modeling is done in two dimensions. Both the water-sand interface and the sand-rock interface are modeled as random rough surfaces following a modified power law spectrum. The rock substrate is assumed to be an elastic solid. Three sediment models are used for the sand layer: the full Biot model for poroelastic media, an effective density fluid model based on the Biot model, and a simple fluid model. The effect of the choice of sediment model used for sand is studied. The finite element results are also compared with perturbation theory and the Kirchhoff approximation in order to further evaluate the validity of considering the underlying interfaces to be flat as a rough sand-rock interface cannot be handled by these models. [Work supported by ONR, Ocean Acoustics.]

8:15

3aUWa2. Modeling range dependent sediment and interface roughness effects on propagation loss with finite elements. Marcia Isakson and Nicholas Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

At the Target and Reverberation Experiment (TREX) off the coast of the Florida panhandle in May 2013, propagation loss was measured over a track with significant sediment and interface roughness variability. In this study, the finite element method is applied to model the effects of sediment and interface roughness variability on transmission loss. Finite elements provide a full wave solution to the Helmholtz equation to the accuracy of the discretization. Therefore, it provides both forward and backward propagation fields. Where available, data will be taken from TREX environment measurements. Additionally, the results will be compared with an energy loss model which relies on the product of the range dependent reflection coefficient. [Work supported by ONR, Ocean Acoustics.]

8:30

3aUWa3. Scattering of sound by a cylindrically symmetric seamount. Ronald Pannatoni (540 Mark Dowdle Rd., Franklin, NC 28734, elliptic@alum.mit.edu)

A three-dimensional waveguide is used to model scattering of sound by a cylindrically symmetric seamount. The waveguide is “closed” in the sense that it has a boundary below the ocean bottom. This makes the spectrum of local wavenumbers discrete. Partial differential equations govern the modal expansion coefficients of the acoustic pressure and of the radial pressure gradient. Their solutions must be finite-valued at the axis of symmetry and satisfy outgoing radiation conditions at the perimeter of the base of the seamount. Direct numerical integration of this problem is unstable, but with variation of parameters and a Riccati transformation an equivalent problem is obtained for which numerical integration is stable. This approach is compared with an alternative [Pannatoni, POMA **14**, 070003 (2011)] that uses leaky modes of an “open” waveguide having no boundary below the ocean bottom. It is shown how coupling between nearest-neighbor modes of the “closed” waveguide relates to the leaky modes of the “open” waveguide. This coupling restricts the step size that numerical integration can use.

8:45

3aUWa4. Computation and accuracy trade-offs in applied reverberation modeling. Anthony I. Eller (OASIS, Inc., 1927 Byrd Rd., Vienna, VA 22182, ellera@oasislex.com) and Kevin D. Heaney (OASIS, Inc., Fairfax Station, VA)

Reverberation modeling and prediction are a research area of intense interest, often involving large, high resolution databases and advanced numerical techniques, and stretching modern computers to their limits. Nevertheless, applications of underwater acoustic systems often need estimates of reverberation sooner than they can be provided by the research community. What results is that applied models for reverberation, in order to meet the pressing needs with available computer memory and allowable computation times, often employ approximations or shortcuts that fall short of current research modeling standards. This paper traces several of the approximations used in reverberation models in the past in order to meet system design and deployment needs. The results show an inherent conflict between basic research and applied interests. The traditional DOD paradigm is that basic research results transition to the applied world. History suggests the opposite, however: Applied research ignores the torpedoes and moves full speed ahead where needed, then forges the chain used to pull reluctant basic research in the needed direction.

9:00

3aUWa5. The research on the coverage area of multistatic sonar under various work modes. Xueli Sheng (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, Heilongjiang, China), Jia Lu (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin Eng. University Shuisheng Bldg. 803, Nantong St. 145, Harbin, Hei Longjiang 150001, China, lujia0507@163.com), Weijia Dong, Jingwei Yin, Longxiang Guo, and Xiaochen Wu (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, Heilongjiang, China)

Absorption loss, which is an important characteristic in the field of underwater acoustic, caused by sea water is often neglected in the research on the coverage area of bistatic sonar. In this article, the coverage area of bistatic sonar considering absorption loss is studied. The change of coverage area caused by absorption loss and the approximation are discussed. When multistatic sonar works in different modes, detection criterion varies, and the coverage area cannot be simply considered as the union set or the intersection of coverage area of each bistatic sonar unit. The definition of multistatic coverage area is discussed, and the coverage area respect to the most relaxed criterion and the strictest criterion is studied.

9:15

3aUWa6. European project for achieving quieter oceans by shipping noise footprint reduction. Christian Audoly and Céline Rousset (CEMIS/AC, DCNS Res., Le Mourillon, BP 403, Toulon 83055, France, christian.audoly@wanadoo.fr)

There is a growing consensus among the scientific community for the need to mitigate underwater noise footprint due to shipping, in order to prevent negative consequences to marine life. In that context, AQUO project started in October 2012, in the scope of the FP7 European Research Framework, for three years duration. The final goal is to provide policy makers with practical guidelines and solutions, acceptable by shipyards and ship owners, in order to mitigate underwater noise due to noise radiation from ships. First, a general presentation of the project will be given, and the logical relationship between the tasks will be outlined. A key element is the development of a "Shipping noise footprint assessment tool," derived from the software Quonops©. This tool can predict the noise map in a maritime area using real time information on ship traffic. Many other studies are undergoing, covering ship underwater radiated noise, propeller noise (including cavitation), and bio-acoustic experiments. The AQUO project team is composed of ship industry, specialized companies, a classification society, research centres and academics, allowing addressing the complexity of the topic. In a second part, some recent results from the project will be given. These will include, for example, the definition of indicators for the impact of shipping noise on marine life, results from vibro-acoustic measurements on several ships at sea, and results from bio-acoustic studies.

9:30

3aUWa7. Mode analysis of distant shipping noise in the SPICE experiment. Mehdi Farrokhrooz, Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., 4450 Rivanna River Way PMB3740, Fairfax, VA 22030, mfarrokh@masonlive.gmu.edu), Matthew A. Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

A major source of ambient noise for frequencies below 100 Hz is due to distant shipping (Wenz, 1962). Shipping noise is trapped in the SOFAR channel and propagates long distances. Since this part of ambient noise excites the low normal modes, it decays with increasing depth. A modal analysis of shipping noise provides valuable insights about its depth dependence. SPICEX offered a unique opportunity to study the ambient noise in the Northeastern Pacific using two vertical line arrays an axial array and a deep array. A 40-hydrophone axial array spanned the SOFAR channel and recorded ambient noise at regular intervals over a year. The axial array data makes it possible to estimate the wavenumber-frequency spectra of ambient noise. We can estimate the mode powers associated with the angle-limited noise in the wavenumber-frequency spectra using a least squares fit. Given the estimated mode powers, depth-dependence of the shipping noise component is predictable. This talk compares the predicted noise levels as a function of depth with data measured by the deep SPICEX line array.

9:45

3aUWa8. Broadband incoherent virtual array beamforming on small scale array for passive detection. Weijia Dong, Xueli Sheng, Chunyan Sun (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., No.145 NanTong St., Harbin 150001, China, messi_881003@163.com), Jia Lu, and Hu Shen (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, Hei Longjiang, China)

It cannot be guaranteed that array spacing corresponding to half-wavelength of each frequency during broadband signals passive detection process. Therefore, for high-frequency, side lobe will rise when array spacing is half the wavelength of the low-frequency. In addition, the main lobe of small-scale array will be wider due to limited array aperture. In order to resolve these two disadvantages for passive detection of small-scale array, virtual array beamforming based on time-delay information is studied in this paper. But this technology often requires high sampling frequency which will increase computational complexity. In order to use virtual array beamforming to processing broadband signals at lower sampling frequency, broadband incoherent virtual array technology based on phase information is introduced in this paper. The effectiveness of these two methods is shown by simulation and sea experiment.

10:00–10:15 Break

10:15

3aUWa9. Alternatives to traditional probabilistic approaches for incorporating environmental uncertainty in ocean acoustics. Steven I. Finette (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

Quantifying environmental uncertainty and its corresponding mapping onto the acoustic field is a subject of considerable importance for underwater acoustics modeling and simulation. The ability to make accurate numerical predictions of basic acoustic quantities, as well as sonar system performance in complex ocean environments depends, in part, on objectively quantifying our imperfect knowledge of the waveguide properties. Over the past few years, a number of approaches have been considered to quantify uncertainty in ocean-acoustics modeling; these methods are grounded in some form of traditional probabilistic reasoning. While probability theory provides a natural framework for representing and quantifying environmental uncertainty, a significant amount of information must be available to specify density functions for the parameters and fields so that the corresponding acoustic field statistics can be computed. This talk will overview alternatives to traditional probability-based reasoning that have been introduced in research areas where conventional specification of probability distributions are not a practical means for describing uncertainty. [Work supported by base funding from ONR.]

10:30

3aUWa10. Spatial uncertainty in higher fidelity acoustic models, and impact on model-data comparisons. Richard L. Campbell, Kevin D. Heaney (OASIS Inc, 11006 Clara Barton, Fairfax Station, VA 22039, campbell@oasislex.com), Phil Abbot, and Chris J. Emerson (OASIS Inc, Lexington, MA)

Acoustic propagation models, such as the parabolic equation (PE), are typically parameterized by a single source depth and one or more receiver depths and ranges. At the higher frequencies, propagation angles, and ranges enabled by advances in computer hardware and refinement of algorithms, the acoustic field predicted by the model can contain features at considerably finer scale than the resolution of the requested output points. The typical practice of simply sampling the field at the output points can introduce artificial large scale structure in the output, due to aliasing. However, averaging out the fine-scale spatial variability can also result in a misleadingly precise value for transmission loss (TL) at a given point. We show that reconciliation of acoustic model predictions with experimental data can be greatly improved by presenting the TL variability due to the expected uncertainty of source and receiver position within an otherwise deterministic model. Even with good estimates of source and receiver positions, this variability can be greater than that due to uncertainty in the model's environmental inputs. Model-data comparisons are demonstrated with examples from a recent ONR effort in the eastern Gulf of Mexico which included controlled TL measurements at 0.4–2 kHz in shallow water.

10:45

3aUWa11. Correlation matrices as a method of source function isolation and classification in ambient noise. Stephen Nichols (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, smn5198@psu.edu) and David L. Bradley (Appl. Res. Lab., The Penn State Univ., State College, PA)

At low frequencies (1–100 Hz), the deep-ocean ambient noise field is a mixture of many source types, including seismic activity, marine life, ship traffic, seismic airgun surveys, and nonlinear surface wave interactions. The location of the sensors used to monitor these types of noise typically result in an extremely dynamic recorded ambient noise field. This study proposes the use of correlation matrices as a tool for identifying the predominant noise sources present in samples of ambient noise. These correlation matrices work by identifying frequency ranges in which the noise levels tend to change at the same time, thus identifying frequency ranges where noise is driven by one particular source mechanism. The effectiveness of this strategy will be demonstrated using low-frequency deep-ocean ambient noise recorded in the Pacific, Atlantic, and Indian Oceans by the Comprehensive

Nuclear-Test Ban Treaty Organization (CTBTO) hydroacoustic monitoring system.

11:00

3aUWa12. Overview of the Target and Reverberation EXperiment 2013 (TREX13). Dajun Tang and Brian T. Hefner (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

The Target and Reverberation Experiment (TREX13) is a shallow water reverberation experiment that endeavored to measure contemporaneously acoustic and adequate environmental data so detailed model/data comparison can be achieved and important environmental factors can be identified for different applications. TREX13 was sponsored by ONR Ocean Acoustics and the SERDP DoD programs. It was conducted during April to June of 2013 off the coast of Panama City, Florida, with participation from multiple institutions and involving three research vessels: The R/V Sharp and R/V Walton Smith from the United States, and the Canadian Force Auxiliary Vessel Quest. From a SONAR viewpoint, reverberation consists of two-way propagation and a single backscatter. Therefore, reverberation, transmission loss, and bottom backscatter were repeatedly measured over a time period of several weeks in the frequency band of 2–10 kHz, along with extensive environmental measurements. Discussed will be planning and execution of the field experiments, strategies and steps for data analysis, and modeling efforts.

11:15

3aUWa13. Solving the inverse problem to extract ship radiated spectra in ocean environments. David P. Knobles (ARL:UT, Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78757, knobles@arlut.utexas.edu)

In ocean acoustics, one is often asked to find the Green's function G from the measured response R in R = GS, where S is the source spectrum. Experiments are often conducted where R is measured and S is known. However, the problem becomes more complicated when S is not known. The application of interest here is the solution for the source spectrum S = Inverse(G) R for a fast moving surface ship in uniform motion in an ocean environment where the propagation includes the interaction of sound with the seabed. What makes the solution possible is the multimodal nature of the propagation over a broadband of frequencies. As an application, the received spectra of merchant ships inside the Reliable Acoustic Path (RAP) range were recorded in the North Pacific Ocean on a vertical line array in an experiment called Church Opal in 1975. The effects of the seabed are clearly evident on the signals received on a hydrophone below the reciprocal depth. As a means of validation, the source spectra solutions of five merchant ships are compared to the spectra of similar ships recorded in other deep-water environments where the effects of the seabed are not present.

11:30

3aUWa14. Utilizing an extended target for high frequency multi-beam sonar intensity calibration. John L. Heaton (Ocean/Mech. Eng., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, jheaton@ccom.unh.edu), Thomas Weber (Ocean/Mech. Eng., Ctr. for Coastal and Ocean Mapping/Joint Hydrographic Ctr., Univ. of New Hampshire, Durham, NH), Glen Rice (NOAA Office of Coast Survey, Ctr. for Coastal and Ocean Mapping/Joint Hydrographic Ctr., Univ. of New Hampshire, Durham, NH), and Xavier Lurton (IMN/NSE/AS, IFREMER, Plouzane, France)

There is an interest in expediting intensity calibration procedures for Multi-Beam Echo-Sounders (MBES) to be used for acoustic backscatter measurements. To this end, a target was constructed of irregularly oriented small chain links arranged in a "curtain" simulating an extended scattering surface, such as the seafloor. Tests with a 200-kHz, 7°, SIMRAD EK60 Split-Beam Echo-Sounder (SBES) were performed in a tank in order to investigate the targets angular- and range-dependent scattering strength. These tests suggest that the amplitude envelope of the scattered signal is Rayleigh distributed and that the backscatter strength depends linearly on the number of active scattering elements. Following these initial validation tests, a 200 kHz Reson SeaBat 7125-SV2 MBES was calibrated in the tank with the same extended target. During the calibration, the MBES was rotated so that every beam was incident on the target. This calibration test was performed once when the target was at normal and once at oblique (45°) incidence. The final output is a "catch all" beam-dependent calibration coefficient, C, determined from the sonar equation.

11:45

3aUWa15. Development of a semi-analytical model for the underwater radiated noise from a driven pile—Comparison of the stationary phase approximation with exact integration for computing an inverse Fourier Transform of vertical wavenumber. Marshall V. Hall (Marshall Hall Acoust., 9 Moya Crescent, Kingsgrove, NSW 2208, Australia, marshallhall@optushome.com.au)

A semi-analytical model is being developed for the noise radiated underwater when a cylindrical pile is struck axially by a hammer. The model is based on the coupled equations of motion for axial and radial vibration of a thin shell. It yields frequency-dependent axial phase velocity and attenuation (due to radiation) and produces a complete description of the shell vibration. For the purpose of describing the total radiated sound pressure, a harmonic solution is obtainable. The "Transform formulation of the pressure field of cylindrical radiators" by Junger and Feit is adopted. The result includes an inverse Fourier transform of a function of vertical wavenumber, and when the integrand is simplified using the stationary-phase approximation, the resulting pressure at short ranges is very large along a downward line in the range-depth plane. The slope of this line is that of the "Mach waves" described by Reinhall and Dahl, and the pressure is many times as great as that of the Mach waves. This approximation is therefore inapplicable. Results are also presented based on using wavenumber integration to compute the Fourier transform.

12:00

3aUWa16. Approximation of a physics-based model of surface reflection loss. Adrian D. Jones, Alex Zinoviev, and David W. Bartel (Maritime Div., DSTO, P.O. Box 1500, Edinburgh, SA 5111, Australia, bearjones@adam.com.au)

A model of coherent acoustic reflection loss at the ocean surface had been prepared by the authors by combining a model of surface roughness loss with a description of surface incidence angle which accounted for the refractive effects of a uniformly stratified distribution of wind-driven bubbles. Here, surface roughness loss was based on a second-order small-slope approximation, and the surface incidence angle was obtained using a formulation for stratified media from Brekhovskikh applied to the sound speed variation resulting from the bubble distribution used by Ainslie [JASA **118**, (2005)]. More recent work by the authors has shown that the analyses for each of surface roughness loss, and surface incidence angle, may be approximated adequately by relatively simple expressions, and that the complete model of surface reflection loss inclusive of the refractive effects of bubbles may be approximated in expressions suitable for hand calculations. Results from the use of this approximated model with a Gaussian-beam acoustic propagation code are compared with results obtained from the authors' more complete model for several surface ducted transmission scenarios. Both sets of results are also compared with predictions based on Monte Carlo parabolic equation (PE) transmission calculations.

Session 3aUWb

Underwater Acoustics and Acoustical Oceanography: Underwater Acoustics and Acoustical Oceanography Poster Session

Gopu R. Potty, Chair
Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882

All posters will be on display from 9:00 a.m. to 12:00 noon, To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10.30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

3aUWb1. On the inversion of sediment density profile by the image source method. Achraf Drira, Laurent Guillon, and Abdel-Ouahab Boudraa (Ecole navale/IRENav, Ecole navale/IRENav, BCRM Brest, CC600, Brest cedex 9 29240, France, achraf.drira@ecole-navale.fr)

Image Source Method is a recently developed method for geoacoustic inversion. Under the Born approximation, the reflection of a spherical wave above a stratified seafloor is modeled as the simultaneous emission of image sources, symmetric of the real one relatively to the layers. These image sources are detected and located by the use of Teager-Kaiser Energy Operator (TKEO), which amplifies sudden changes in signal amplitudes, and by a triangulation scheme. Time and arrival angle of the recorded signals coming from the image sources are the input values of the inversion algorithm which gives the thickness and sound speed of each detected sediment layer by the use of Snell-Descartes laws. The objective of the present work is to extend this method to the inversion of the sediment density profile. To this end, one supplementary data is required. Experimental amplitudes corresponding to arrival times of the reflected signals, detected by TKEO, are compared to theoretical ones computed by a numerical evaluation of the Sommerfeld integral. Having first inverted sound speed profile and neglecting absorption coefficient, density is the only remaining parameter and thus can be obtained. The effectiveness of this method is tested on both synthetic and real data.

3aUWb2. Statistical inference of seabed sound-speed structure in the Gulf of Oman Basin. Jason D. Sagers and David P. Knobles (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu)

Addressed is the statistical inference of the sound-speed depth profile of a thick soft seabed from broadband sound propagation data recorded in the Gulf of Oman Basin in 1977. The acoustic data are in the form of time series signals recorded on a sparse vertical line array and generated by explosive sources deployed along a 280 km track. The acoustic data offer a unique opportunity to study a deep-water bottom-limited thickly sedimented environment because of the large number of time series measurements, very low seabed attenuation, and auxiliary measurements. A maximum entropy method is employed to obtain a conditional posterior probability distribution (PPD) for the sound-speed ratio and the near-surface sound-speed gradient. The multiple data samples allow for a determination of the average error constraint value required to uniquely specify the PPD for each data sample. Two complicating features of the statistical inference study are addressed: (1) the need to develop a cost function that can both utilize the measured arrival structure and mitigate the effects of data errors and (2) the effect of small bathymetric slopes on the structure of the bottom interacting arrivals. [Work supported by ONR.]

3aUWb3. Issues in estimating the seafloor scattering cross section with synthetic aperture sonar. Derek R. Olson (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, olson.derek.r@gmail.com) and Anthony P. Lyons (Appl. Res. Lab., Penn State, University Park, PA)

The high resolution capabilities of synthetic aperture sonar (SAS) make it an attractive technology for estimating the seafloor scattering cross section. However, differences between traditional systems used for backscattering measurements, whose apertures span a few wavelengths, and synthetic arrays, whose apertures can span thousands of wavelengths, can violate the assumptions used in typical cross section estimation techniques. The issues introduced by very long synthetic arrays are caused by the integration of acoustic energy over a range of azimuth and grazing angles in the local coordinates of the resolved area on the seafloor. The resulting pixel intensity is not directly proportional to the scattering cross section, and methods other than solving the sonar equation are required. This research explores the effect of naively using SAS data to estimate the cross section and presents alternative estimation techniques.

3aUWb4. Estimation of seabed bottom loss using wideband signal for underwater acoustic communication. Sung-Hoon Byun, Sea-Moon Kim, and Yong-Kon Lim (Ocean System Eng. Res. Div., KRISO, 171 Jang-dong Yeseong-gu, Daejeon 305-343, South Korea, byunsh@gmail.com)

Performance of acoustic communication through shallow underwater channel is affected by surface and bottom interaction, and the bottom loss has great importance for long-range signal transmission. In 2013, KRISO has performed a shallow-water communication experiment near Jeju island, South Korea, and the experiment aims at evaluating communication link budget as well as measuring the channel fading statistics. The measured sound velocity profiles show the existence of strong thermocline resulting in downward refraction and therefore its long-range signal propagation is strongly dependent on the seabed characteristics. We use a wideband signal, which was originally designed for channel impulse response estimation. The signals were recorded using a four-element receiver array with two different source depth configurations which enable observing the seabed at various grazing angles. The estimated bottom loss is used to estimate the transmission losses at longer ranges, and they are compared with the measured values to examine the estimated bottom loss.

3aUWb5. Simulation and analysis of acoustic impedance measurement techniques for marine sediments using sonar. João P. Ristow (Mech. Eng., Federal Univ. of Santa Catarina, Rua Lauro Linhares, 657, Apto. 203B, Florianápolis, Santa Catarina 88036-001, Brazil, jpristow@gmail.com), Guillaume Barrault (Oceanogr., Federal Univ. of Santa Catarina, Florianápolis, Santa Catarina, Brazil), Julio A. Cordioli, Gregório G. Azevedo (Mech. Eng., Federal Univ. of Santa Catarina, Florianápolis, Santa Catarina, Brazil), Antônio H. Klein, and Marina Bousfield (Oceanogr., Federal Univ. of Santa Catarina, Florianápolis, Santa Catarina, Brazil)

The characterization of seabed sediments using acoustics techniques is a wide field of research. A key feature in the sediment characterization is the estimation of its acoustic impedance, from which density and wave propagation speed in the sediment can be derived. The sea bottom acoustic impedance is a property that rules the backscattering strength, together with other sediment features such as slope and roughness. In this sense, sonar devices can be used to measure the backscattering strength of a material lay up on the sea bottom and estimate the aforementioned sediment features. However, one of the main difficulties is to obtain sufficient complementary information in order to decompose the backscattering strength of a giving sea bottom and obtain its proper acoustic impedance. Hence, the aim of this work is to evaluate and compare different techniques for measurement and signal processing that will be used to indirectly calculate the sediment's acoustic impedance from numerically simulated sonar data.

3aUWb6. Comparison between the Biot-Stoll model, the grain-shearing model, and the effective density fluid model with respect to sediments in the Yellow Sea. Eunghwy Noh, Hunki Lee (School of Mech. Eng., Yonsei Univ., Eng. Bldg. A117, 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, hwyya@yonsei.ac.kr), Oh-Cho Kwon (The 6th R&D Inst. Naval System Directorate Principal Researcher, Agency for Defense Development, Changwon, South Korea), and Won-Suk Ohm (Mech. Eng., Yonsei Univ., Seoul, South Korea)

There are some distinctive geophysical properties of sediments in the Yellow Sea, where mudflats are developed extensively. To select a physical model for acoustic wave propagation in the sediments is a critical issue in the analysis and the conceptual design of buried object scanning sonars. We consider three different sediment models to describe physical and geoacoustic properties of the sediments in the Yellow Sea: the Biot-Stoll model, the grain-shearing model, and the effective density fluid model (EDFM). These sediment models are evaluated in terms of their predictions of sediment properties compared with the measurement data. [This work was conducted in the Unmanned Technology Research Center (UTRC) sponsored by the Defense Acquisition Program Administration (DAPA) and the Agency for Defense Development (ADD) in the Republic of Korea.]

3aUWb7. Propagation of nonlinear sound beams in marine sediments. Hunki Lee, Eunghwy Noh (Mech. Eng., Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, ssevi2@yonsei.ac.kr), Oh-Cho Kwon (The 6th R&D Inst. Naval System Directorate Principal Researcher, Agency for Defense Development, Changwon, South Korea), and Won-Suk Ohm (Mech. Eng., Yonsei Univ., Seoul, South Korea)

Parametric arrays are recently considered as directional sound sources for locating buried objects in marine sediments. To understand how nonlinear sound beams generated by parametric arrays propagate in marine sediments, a theoretical framework that accounts for the combined effects of diffraction, absorption, nonlinearity, and the physics of sediment is necessary. In this paper, the KZK equation is extended to augment a number of well-known sediment models. Nonlinear evolution of sound beams are computed and compared under different sediment models, namely, the fluid model, Buckingham's grain-shearing model, the Biot-Stoll model, and the effective density fluid model (EDFM). [This work was conducted in the Unmanned Technology Research Center (UTRC) sponsored by the Defense Acquisition Program Administration (DAPA) and the Agency for Defense Development (ADD) in the Republic of Korea.]

3aUWb8. Broadband modeling of sound propagation in shallow water with an irregular elastic bottom. Li Li (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Shuisheng Bldg. Rm. 1303, Nantong St., No.145, Nangang District, Harbin, Heilongjiang 150001, China, liliandxy@126.com), Shengchun Piao, Haigang Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China), Yaxiao Mo, and Jun Tang (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, China)

The purpose of this work is to study the propagation of broadband sound pulses in shallow water environments. It is essential for an underwater pulse propagation model to handle bottom interaction, range-dependence, and wide-angle propagation in shallow water. Therefore, a comparatively realistic model that consists of a fluid overlying an irregular elastic bottom is treated, where the effects of shear wave are included. The range-dependent seismo-acoustics problem in frequency-domain is solved by a parabolic equation model. Fourier synthesis of frequency-domain solutions is implemented to model the received time series of a broadband sound propagation. And parallel programming is tried to improve the computational efficiency. Dispersion characteristics are exhibited by multiple mode arrivals during the propagation, including the dispersion of normal modes and mode 0 (the Scholte wave). The dispersion analysis of normal modes and the Scholte wave are demonstrated under different types of elastic bottoms. Energy converting between trapped modes and leaky modes due to slope of bottom is also analyzed.

3aUWb9. Laboratory measurements of high-frequency, broadband acoustic scattering of growing sea ice and oil beneath sea ice. Christopher Bassett, Andone C. Lavery, and Ted Maksym (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 12, Woods Hole, MA 02543, cbassett@whoi.edu)

The morphology of sea ice during the early stages of growth is strongly dependent on environmental conditions. Under calm conditions, congelation ice forms through downward growth of ice crystals from the water surface. Under turbulent conditions (surface waves), rapid freezing of ice crystals occurs in the upper water column (frazil ice), eventually consolidating into pancake ice through repeated collisions and agglomeration of the loose frazil crystals. It is expected that high-frequency scattering from the basal layer of the ice varies for different sea ice types and can reveal structural information that governs the behavior of the ice and its interactions with the environment. Broadband scattering measurements of sea ice are presented beginning with ice-free conditions and through initial stages of growth in laboratory experiments for both congelation and frazil ice. With increased interest in drilling for hydrocarbon resources in the Arctic and the associated environmental concerns of an oil spill in ice-covered waters, improved methods for detection of crude oil both under or frozen within sea ice are needed. Acoustic scattering data are presented demonstrating how the scattering changes when crude oil is spilled beneath the ice.

3aUWb10. Detectability of navigation obstacles with forward looking sonar in presence of boat wakes: Observations in Narragansett Bay. Alexander M. Yakubovskiy, Nabin S. Sharma, and Matthew J. Zimmerman (Signal Processing, Farsounder, Inc., 151 Lavan St., Warwick, RI 02888, alex.yakubovskiy@farsounder.com)

A Forward Looking Sonar (FLS) is designed to detect obstacles ahead of a vessel. Detection performance of an FLS is affected by several environmental factors surrounding the sonar. Wakes generated by nearby vessels degrade the detection capability of an FLS. With more than 40 000 registered boats, Rhode Island's Narragansett Bay is a good example of a zone with heavy boat traffic where the performance of an FLS may be effected. FarSounder has collected vast amounts of data while testing and demonstrating its FLS sonars in Narragansett Bay. Using that data, this paper presents observations of FLS performance with and without boat wakes. In order to explain the sonar performance, a probability of wake presence in sonar field of view is estimated based on shipping density, wake size, and lifetime.

3aUWb11. Parameter measurement research of sonar echo highlights. Cheng He, Anbang Zhao, Bin Zhou, Xuejing Song, and Fang Niu (College of Underwater Acoust., Harbin Eng. Univ., Heilongjiang Province,Harbin City, Area Nangang, Nantong St. No.145, Shuisheng Bldg., Harbin, Heilongjiang 150001, China, hecheng@hrbeu.edu.cn)

Highlights sonar model provides an important basis for complex target modeling. In this paper, a parameters measurement method for the far-field sonar target highlight is proposed, time reversal mirror (TRM) technology and transponder technology combined with this method. First, get the number of target highlights. Then, put transponders with the same number along underwater the target which is measured, those transponders must be close at each highlight. A wideband signal is transmitted at measuring position which is far away from target, those transponders at target side answer one by one. Record the echoes coming from target and transponders. Time-delay, amplitude, and phase-jump of highlights can be calculated by processing those echoes in TRM and matrix operations methods. It is found through simulation by MATLAB that this target highlight algorithm can accurately measure the three important parameters of highlight model in negative SNR. By getting those parameters accurately, some other parameters can be calculated, for example, target strength. Target echo can also be predicted with those parameters, it will be very useful in target strength measurement and active target stealth technology then.

3aUWb12. The instantaneous frequency variance feature of underwater bottom target. Xiukun Li and Zhi Xia (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145, Nantong St., Rm. 305, Shuisheng Bldg., Nangang District, Harbin, Heilongjiang 150001, China, lixiukun@hrbeu.edu.cn)

The key of underwater target recognition is extracting stable target feature from the complicated mixed signal. Deal with LFM pulse, the time-frequency character of target echo is fixed and linear; however, the time-frequency feature of target echo is sensitive with the SRR, the angle of incidence wave, and the shape and the material of target. Based on the geometric acoustic scattering character of a classical bottom target model, the time-frequency distribution of target echo is treated as a bidimensional image in this paper. During with image rotation, the time-frequency distribution of target echo is transformed to a line spectrum, and then, the instantaneous frequency variance feature is extracted. The experiment data processing result has shown that the clusters of target echo and reverberation on this feature are stable for various environments and the shape and the material of targets. The method proposed in this paper is meaningful for construct a universal feature space in the detection of underwater bottom target.

3aUWb13. Spatial diversity in application of the decomposition of the time reversal operator (DORT) method to imaging of an extended target. Chunxiao Li, Mingfei Guo (MOE Key Lab. of Mech. Manufacture and Automation,College of Mech. Eng., Zhejiang Univ. of Technol., 18# Chao-Wang Rd., Zhejiang, Hangzhou 310014, China, chunxiaoli@zju.edu.cn), and Qianliu Cheng (Hangzhou Appl. Acoust. Res., Hangzhou, China)

Taking advantage of the multipath propagation in shallow water waveguide, high resolution can be achieved with the decomposition of the time reversal operator method. In this paper, we investigate how to use the spatial diversity for imaging of an extended target. It is assumed that the target is composed of an infinite number of random, isotropic, and independent scatterers, uniformly distributed over an unknown region. The eigenvalues and eigenvectors associated with the target are first determined. We then employ the frequency characterizations of scatterers from a singular value decomposition of the time reversal operator to localize and image the target. Tank experiments are carried out for a steel cylinder in two kinds of random media: a shallow water waveguide with multiple scatterers and without scatterers. The effects of the random media on the choice of the parameters of the signal processing are also studied.

3aUWb14. A possibility to use respiratory noises for detecting diver and monitoring his physiologic status. Sergei Gorovoy (Far Eastern Federal Univ., Vladivostok, Russian Federation), Vladimir Korenbaum, Aleksandr Tagiltcev, Anatoly Kostiv (Pacific Oceanologic Inst., 43, Baltiiskaya Str., Vladivostok 690041, Russian Federation, v-kor@poi.dvo.ru), Aleksey Borodin (Appl. Problems Section FEB RAS, Vladivostok, Russian Federation), Irina Pochekutova, Aleksey Vasilistov, Aleksandr Krupenkov (Pacific Oceanologic Inst., Vladivostok, Russian Federation), Denis Vlasov (Far Eastern Federal Univ., Vladivostok, Russian Federation), and Anton Shiryaev (Pacific Oceanologic Inst., Vladivostok, Russian Federation)

The objective is acoustic detection of submerged diver and monitoring his physiologic condition. A possibility to use diver's own respiratory noises for this aim is analyzed. Respiratory noises were recorded above trachea of submerged scuba diver and in the water layer of shallow-water bay. Both signals contain quasi-periodic components induced by amplitude modulation of wideband respiratory noises with the rate of breathing maneuvers. These components are detected in water layer by means of energy processing of single hydrophone response (in a frequency band of 200–500 Hz) at the distances up to 50 m. The breathing rate is estimated by means of spectral transform of the signal envelope. It is well known that this index represents human individual physiologic status. Thus, it is pertinent for submerged diver's condition monitoring. The quasi-periodic components if detected may be used to estimate time delays in responses of several hydrophones, for example, by means of the signal envelopes correlation processing. These time delays are pertinent to find diver's location by triangulation technique.

3aUWb15. Active acoustic detection of subsea oil and gas leaks; model prediction and measurements. Geir Pedersen, Rune Hauge (Instrumentation, Christian Michelsen Res. AS, P.O. Box 6031, Bergen 5892, Norway, geir.pedersen@cmr.no), and Rokas Kubilius (Ecosystem Acoust., Inst. of Marine Res., Bergen, Norway)

There is a need in the oil and gas industry for technology for real time monitoring of subsea structures. National research strategies emphasize the need to develop innovative solutions to detect, contain, and clean up spills. The technology should ideally cover a large area and be able to detect leaks up to 500 m from the sensor. As part of a project where a leak detection technology based on active acoustics is developed, effective medium models have been used to predict the acoustic signal from small oil and gas leaks. Model results were compared with multifrequency data from controlled *in situ* measurements of oil and gas releases, collected with scientific echosounders. The measurements also include stereo camera images for bubble/droplet and plume size estimation. The models generally predict the measurements within 3 dB, when taking into account the variability in plume dimensions and bubble sizes the difference is smaller.

3aUWb16. On marine mammal acoustic detection performance bounds. Yin Xian and Loren Nolte (Elec. and Comput. Eng., Duke Univ., 857 Louise Circle, Durham, NC 27705, poline3939@gmail.com)

Most current research on marine mammal acoustic detection, classification, and localization does not consider optimizing detector algorithm performance for marine mammal vocalizations that have been propagated through an uncertain multipath environment. In this study, a Bayesian likelihood ratio approach, using the original time series data, is used to benchmark optimal detection performance (ROC's) for a known marine mammal source vocalization propagated through both a known and an example uncertain ocean environment. In addition, for these same ocean environments, the detection performance (ROC's) is obtained for several algorithms based on the spectrogram of the original data. Since the spectrogram does not preserve detailed phase information contained in the original data, any algorithm based on the spectrogram is not likely to be optimum for detection. The initial results show the additional detection performance gain possible over spectrogram based algorithms. Simulations and preliminary detection performance results (ROC's) are presented using a mathematical model from the literature of the North Atlantic Right Whale (NARW), along with numerical acoustic propagation software.

3aUWb17. Broadband class I flextensional transducer. Yu Lan, Wei Lu, Yongjie Sang, and Tianfang Zhou (Harbin Eng. Univ., Nantong St. No.145, Hei Longjiang Province, Harbin, Harbin 0086, China, lanyu_2013@126.com)

Researches about high-power, low-frequency projectors are the most important technology for long range sonar systems. Flextensional transducer is a typical low-frequency, high-power projector, which has the advantages of small volume and light weight due to displacement amplification. In this paper, attention is focused on expanding bandwidth of the flextensional transducer. Class I flextensional transducer usually consists of a slotted convex shell and a driving stack. There are two kinds of main vibration modes at low frequency, the first flexural mode and the membrane mode. They can be coupled and broadband transmitting response was obtained. Finite element (FE) techniques are applied to the design of this multiple-mode-coupling transducer. Finite element model was made with ANSYS software and it can be used to do vibration mode analyze, operative mode simulation and structure optimization. Base on the analyze results, the multiple-mode-coupling flextensional transducer was designed and tested. Key words: flextensional transducer; broadband; multiple-mode-coupling; Finite Element Method

3aUWb18. Doubly resonant underwater acoustic transducer for long distance sound propagation. Andrey K. Morozov (Teledyne, 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com)

The presentation describes the design and test of innovative deep water low frequency sound source for long range acoustic communications and navigation. The light-weight, low frequency (200–1000 Hz), broadband underwater sound source comprises an inner resonator tube with thin walls tuned to a certain frequency surrounded by a shorter, larger-diameter, lower frequency tuned outer resonator tube. These resonating tubes are opened on both ends and made of carbon fiber. The tubes are asymmetrically shifted along the main axis and sound pressure can penetrate from internal pipe though the area under the shifted external pipe into that external pipe and back. By changing length of shifted area the coupling coefficient of two resonators can be regulated to achieve a necessary bandwidth. The design uses depth independent oil-filled acoustic transducer. The transducer is light, operational for all ocean depth, and reasonably broadband. A multiresonant systems usually need a precise complicated adjustment of their parameters to get necessary bandwidth with limited variability of frequency response inside the frequency range. It can be even more complicated, when design includes new, never tested before, materials. Application of finite element analysis allowed to predict necessary parameters and avoid a long series of water tests with parameters adjustment.

3aUWb19. Optimizing eigenray generation for real time simulations. Andrew J. Fredricks (Undersea Warfare Weapons, Vehicles, and Defensive Systems, Naval Undersea Warfare Ctr., 487 NewBedford Rd., Rochester, MA 02770, fredra@comcast.net) and Sheri Martinelli (Undersea Warfare Weapons, Vehicles, and Defensive Systems, Naval Undersea Warfare Ctr., Newport, RI)

Real time simulation is a critical supplement to in-water testing for the assessment of naval system performance. Limitations exist on the number of independent runs, on geographical locations in which in-water tests can be performed, and on the amount of control experimenters possess over the test environment. Many systems have critical time-sensitive functionality (e.g., acoustic homing) which constrains the ability to produce realistic time series for injection; but a reduced fidelity solution can still be of use. Graphics hardware (GPU) has become a significant computing platform in its own right. Its application requires a mapping of the propagation algorithm to the GPU computing paradigm and careful tweaking to squeeze out maximum performance. We will look at taking from theory just what we need to hand-tune code for a GPU + CPU computing platform, and the limitations of a high speed, range dependent, eigenray code. We also consider a related approach that uses the resulting eigenrays to initialize an iterative method which updates the eigenray solution as the source and receiver update their relative positions. [Work funded by the Office of the Secretary of Defense, Test Resource Management Center's Resource Enhancement Project element of the Central Test and Evaluation Investment Program.]

3aUWb20. Acoustic propagation in surface channel formed by low salinity water in the East China Sea and the tropical Atlantic Ocean. Juho Kim, Hansoo Kim, Dong-guk Paeng (Dept. of Ocean System Eng., Jeju Natl. Univ., Ara 1 dong, Jeju 064-756, South Korea, lizard@jejunu.ac.kr), and Jongkil Lee (Mech. Eng. Education Dept., Andong Natl. Univ., Andong, South Korea)

Salinity is usually neglected in underwater acoustics because of its minimal contribution on sound speed variation. However, it is required to consider in calculation of sound speed and acoustic propagation for low salinity water freshened by continental run-off. Furthermore, the acoustic characteristics of low salinity environment are not fully investigated yet. Therefore, regional difference of acoustic propagation in low salinity water was studied by comparing the acoustic characteristics of the East China Sea with those of the tropical Atlantic Ocean in this paper. Frequency dependency of sound propagation in the haline channel was analyzed with transmission losses and low frequency cut-off using oceanic data from NODC (National Oceanographic Data Center). The tropical Atlantic Ocean showed larger channel depth, critical angles and less transmission loss in the haline channel than the East China Sea. The cut-off frequency in the haline channel was computed as around 1 kHz and 5 kHz in the tropical Atlantic Ocean and the East China Sea, respectively. The effects of low salinity water on sound propagation showed regional characteristics, and need to be considered in sonar operation near sea surface. [This work was supported by Defense Acquisition Program Administration and Agency for Defense Development under the contract UD130007DD.]

3aUWb21. Simulation of underwater environments using the Discrete Huygens Modeling. Renato T. de Carvalho, Gregário G. Azevedo, and Júlio A. Cordioli (Mech. Eng., Universidade Federal de Santa Catarina, Campus Universitário Reitor João David Ferreira Lima, Florianápolis 88040-900, Brazil, renato@lva.ufsc.br)

The Discrete Huygens Modeling (DHM) is a numerical method that applies the Huygens principle to a discretized medium in order to provide time domain solutions to wave propagation problems. The method was originally conceived and applied in Electromagnetism, but previous works have shown that DHM is also a viable and promising approach for the simulation of acoustic problems. The main advantages of the method are its reduced computation cost when compared with other numerical methods and its relatively easy implementation and parallelization. However, little can be found in the literature about the use of DHM to underwater acoustic problems, and previous work have been limited to acoustic propagation only in the water column. In this work, a simple model of the acoustic field generated in the water column by a monopole was created, and the prediction of transmission, reflection, and scattering processes at the interface with the sediment were carried out. A DHM code has been fully implemented allowing the modeling of multiple media. The results obtained with the DHM model were compared with a Finite Element model yielding very good agreement, while the DHM approach proved to be much faster.

3aUWb22. An error reduction method for double-plane nearfield acoustic holography based on boundary element method. Dejiang Shang and Yongwei Liu (College of Underwater Acoust. Eng., Harbin Eng. Univ., Rm. 1205, SHUI SHENG Bldg., 145 Nantong St., Nangang District, Harbin 150001, China, shangdejiang@hrbeu.edu.cn)

Nearfield planar acoustic holography method based on BEM is often used in experiments because it is very convenient to scan the sound field on planes. However, the sound field reconstruction error on the sound source surface is usually very big due to the dimension limitation of measuring planes. The method proposed here, by measuring some extra far field points out of the planes, which are combined with those on the planes to reconstruct the sound field on the source surface, can reduce this sort of error. The numerical simulation has been carried out by this method for a spherical sound source with radius of 1.0 m. The double planes are located at $x=\pm 0.6$ m, assumed that the sound source is at (0,0,0) in Cartesian coordinate. The max reconstructing error of sound pressure on the sound source surface can be reduced from 6 dB to less than 2 dB at frequency 500 Hz when we add two extra far field points at ($0,\pm 10$ m,0). Obviously, this method is very meaningful in practical experimental tests. More numerical simulation by

this method for more complex sound source field reconstruction will be shown in the full paper.

3aUWb23. An equivalent physical model of three acoustic waves interaction in nonlinear medium. Desen Yang (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, China), Haoyang Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145 Nantong St., Harbin 150001, China, zhanghaoyang@hrbeu.edu.cn), Shengguo Shi (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, China), Wei Jiang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China), and Jie Shi (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, China)

Among the parametric effects, three waves interaction has been widely investigated in many fields of physics, such as nonlinear optics and plasma. The interaction of three acoustic waves in the medium with quadratic nonlinearity, which must satisfy conditions on their frequencies and wavenumbers, is essentially confined to energy exchange between three components. Based on Burgers equation, the propagation rules of three acoustic waves, which are called a resonant triad, are governed by a drastically simplified system of three coupled nonlinear ordinary differential equations. A mathematical equivalence between the equations for an acoustic triad and a simple parametric vibration system, the undamped elastic pendulum, is discussed in this paper by a multiple time-scale analysis. We study the dynamics of this system, drawing analogies between its behavior and that of the acoustic triad. Finally, it is certified that three acoustic waves interaction can be described by Mathieu type equations in case one acoustic wave is much stronger than the others in three waves. This means that experiments with an elastic pendulum can give us new insights into dynamics of mechanism in three acoustic waves interaction.

3aUWb24. Theory and experiment of the resonance sound wave interactions in water. Desen Yang (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, China), Wei Jiang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Heilongjiang, Harbin HL 451, China, jiangwei_first@126.com), Shenguo Shi (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, China), Haoyang Zhang, and Shiyuan Jin (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China)

Equations governing nonlinear interactions among acoustic waves in water which satisfy the resonance conditions are derived from the Burgers equation. Taking into account of the second-order nonlinear, the three-wave interaction is the fundamental process of the interaction equations. Then, taking the three-wave interaction equation as an example, the energy transfer mechanism among the three acoustic waves is quantitatively analyzed. And the three-wave sound energy propagation is studied through numerical calculation. An interesting phenomenon is found that, when the three acoustic waves meet the relationship of a weak low-frequency sound and two strong high-frequency sounds, the energy of low-frequency sound will be amplified or reduced in some regions during the three-wave propagation. And the location and size of the regions are affected by the acoustic wave amplitude, frequency, and phase. The variation severities of low-frequency acoustic wave energy are mainly determined by other two high-frequency acoustic wave energy. The frequencies of two high-frequency sound waves have little effects on the energy of low-frequency sound wave. The region whether is amplification or reduction is determined by the phase difference of three waves. The variation laws of low-frequency sound energy are also verified by the experimental results in river.

3aUWb25. Predicting range-frequency interference patterns from broadband sources in a range-dependent, continental shelf environment. Alexander W. Sell (Acoust., Penn State Univ., 830 Cricklewood Dr., Apt. 207, State College, PA 16803, aws164@psu.edu) and R. Lee Culver (Acoust., Penn State Univ., University Park, Oregon)

The 2007 CALOPS dataset contains horizontal line array received signals from numerous surface vessels transiting an area off the coast of southeast Florida. Parabolic equation propagation models using measured bathymetric, sound velocity profile, and sediment parameter data suggest that substantial mode coupling should be occurring for several down-slope propagation scenarios in the dataset. However, received acoustical data for these scenarios, in the form of range-frequency interference patterns, are best described by adiabatic propagation. This discrepancy has implications that affect waveguide invariant parameter estimation, which is necessary for invariant-based passive ranging. The causes of the incongruous model output will be discussed, as well as their effects on estimating waveguide invariant distributions for these scenarios. [This research was supported by the Applied Research Laboratory, at the Pennsylvania State University through the Eric Walker Graduate Assistantship Program.]

3aUWb26. Coherence measurements of acoustic normal modes during one month of internal wave events on the New Jersey continental shelf. Lin Wan and Badiey Mohsen (College of Earth, Ocean, and Environment, Univ. of Delaware, 003 Robinson Hall, Newark, DE 19716, wan@udel.edu)

During the Shallow Water Acoustic Experiment 2006 (SW06) conducted on the New Jersey continental shelf, the three-dimensional (3D) temperature field for one month of internal wave (IW) events has been reconstructed by Badiey *et al.* [J. Acoust. Soc. Am. El. **134**(5), 4035 (2013)]. The aforementioned IW events, with the angle between the acoustic track and the IW front varying from -8° to 83°, were measured while simultaneously acoustic signals were transmitted from fixed sources at an alongshelf distance of about 20 km with frequencies at 87.5–112.5 Hz (m-sequence), 175–225 Hz (m-sequence), 270–330 Hz (chirp), and 470–530 Hz (chirp), respectively. The acoustic signals were recorded by an L-shaped hydrophone array moored inside the area with IW measurements. The main goal of this paper is to analyze the coherence of acoustic normal modes accompany with the simultaneously measured IW events. The coherence of acoustic normal modes decomposed from the measured acoustic field is obtained as a function of frequency and IW parameters, such as the IW propagation direction, amplitude, coherence length, etc. The relationship between the modal coherence and IW parameters is discussed. [Work supported by ONR322OA.]

3aUWb27. Fluctuations of arrival time and frequency shifts of a wideband sound signal in the presence of coastal internal Kelvin waves in shallow water. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru) and Andrey Lunkov (Wave Res. Ctr., Geneal Phys. Inst., Moscow, Russian Federation)

Internal Kelvin waves (IKWs) in a near shore area of shallow water with a curvilinear coastal line (gulf, lake, and channel) where the radius of curvature can be about 5–30 km are considered as a reason for the sound field variability. Amplitude of the thermocline vertical displacement due to IKW in such an area changes along the distance from the center of this circle (minimal value, about null) to the coast (maximal value, up to 5 m). Numerical simulations are implemented to study the effect of IKWs on the propagation of acoustic waves generated by a wideband (0.7 to 1 kHz) sound source. The results of the simulations show that the signal propagation time changes up to 1% following the thermocline displacement caused by IKW. Also, considerable shifts (more than 30% of the central frequency) of the interference pattern in the frequency domain are observed. The shifts change with distance from the shore and have opposite directions for positions of a single hydrophone receiver above or below the thermocline. Analytical estimates are obtained; experimental setup and a possible scheme of monitoring IKW are discussed. [Work was supported by RFBR and BSF.]

3aUWb28. Influence of short time scale water column fluctuations on broadband signal intensity. Justin Eickmeier and Badiey Mohsen (CEOE, Univ. of Delaware, 17 McCord Dr., Newark, DE 19713, jeickmei@udel.edu)

The KAM11 experiment was conducted in 100 m of water off the Western side of Kauai, Hawaii, in June/July, 2011 during which two identical bottom mounted tripods, separated by 1 km, transmitted reciprocal chirp sequences. A monitoring hydrophone was suspended at a depth of 25 m from the *R/V Kilo Moana* approximately midway between the tripod

stations. Impulse response measurements at the suspended hydrophone show intermediate arrivals between direct-path/bottom-bounce and surface-bounce arrivals unique to only one tripod source. Focusing and defocussing of the intermediate arrivals show the influence of fluctuations in the sound speed profile over time scales from seconds to hours. Short term intensity variations arise from surface driven vertical undulations in the water column. Over longer time scales changes in the thermocline govern the evolution of the intermediate arrival. Data model comparison is conducted with ray tracing and parabolic equation modeling. [Work supported by ONR 322OA.]

3aUWb29. Model and data comparisons of ocean acoustic intensity statistics in the Philippine Sea 2010 experiment. Andrew A. Ganse, Rex K. Andrew, Frank S. Henyey, James A. Mercer (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aganse@apl.washington.edu), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

The statistics of intensity fluctuations in dual-band, low-frequency, ocean acoustic transmissions across 500 km in the Philippine Sea 2010 experiment appear to be at odds with the standard micro-multipath theory predicting them. M-sequence encoded acoustic signals at 200 Hz and 300 Hz were transmitted for 54 h from a ship-suspended multiport source to a distributed vertical line array (DVLA) with 149 working hydrophones covering most of the 5500 m water column. Histograms of the received intensity fluctuations are approximately exponential. Intensity fluctuations at the two frequencies are strongly correlated. Scintillation indices and variances of log intensity are high ($>$1 and $>$5.57dB, respectively), suggesting the measurements are not yet in a saturated regime but still significantly scattered. However, for all this, the pulses received at the DVLA are virtually the same width as those recorded next to the source; no pulse spreading or arrival splitting is seen. So, the fading observed in the data does not appear to match this causal mechanism. To explore the connection with micro-multipath theory more fully, we compare the intensity statistics from the Philippine Sea 2010 data with those calculated both from simulated random oceans and from Flatté and Rovner's path integral theory. [Work supported by ONR.]

3aUWb30. Stochastic characterization of acoustic signals for sequential dispersion tracking and geoacoustic inversion. Nattapol Aunsri and Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu)

In previous work, we had shown how sequential Bayesian filtering methods can be used for the successful extraction of dispersion curves from broadband long-range acoustic data. Here, we extend this work by tracing carefully the true nature of the noise and the resulting probability density observations of the spectrogram of the received time series, employed in the tracking. The Gaussian model typically used in instantaneous frequency tracking relies on the assumption that noise is additive in the frequency domain. This model is, however, inaccurate. We discuss a chi-squared model of the acoustic data perturbations and its role in dispersion curve tracking. The new method provides much clearer curves than those computed with previous approaches. We demonstrate the potential of the technique by applying it to synthetic and real data for dispersion curve estimation and bathymetry and sediment sound speed inversion. [Work supported by ONR.]

3aUWb31. Low frequency sound absorption in the Arctic Ocean: Potential impact of ocean acidification. David Browning (139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com), Peter D. Herstein (24 Mohegan Rd., Charlestown, RI), Peter M. Scheifele (Dept. of Comm. Sci., Univ. of Cincinnati, Cincinnati, OH), and Raymond W. Hasse (145 Old Colchester Rd., Quaker Hill, CT)

The principal mechanism for low frequency absorption in seawater is a boron reaction that is *p*H dependent; the lower the *p*H, the lower the absorption. Twenty seven years ago, Mellen *et al*. [J. Acoust. Soc. Am. **82**, S30 (1987)] computed the low frequency sound absorption for the Arctic Ocean. Since the time the carbon dioxide (CO_2) level in the atmosphere has been continually increasing. Experts predict that the resulting ocean acidification may increase by up to 170% this century. Acoustically, the Arctic Ocean is most sensitive to rapid change not only because the cold water readily absorbs CO_2, but also because the sound channel axis is at or near the surface. The range of reduction in the low frequency sound absorption is presented based on possible future acidification scenarios, mindful that this is just one component of a complex evolution that is occurring in the Arctic.

3aUWb32. Sensing stratified turbulence and shear instability using an acoustic high frequency broadband backscatter array. Jonathan R. Fincke (AOSE, MIT/WHOI Joint Program, 99 Hancock St., Apt. 10, Cambridge, MA 01239, jfincke@mit.edu) and Andone Lavery (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA)

High-frequency broadband acoustical backscattering measurements with an array of transducers have allowed the temporal and spatial evolution of shear instabilities in a strongly stratified estuarine environment to be investigated. Development of accurate remote sensing techniques for estimating mixing are of significant interest to the geophysical fluid dynamics community. An array of six high-frequency (120–600 kHz) broadband and narrow beam width (1 to 6 degrees half-beamwidth, depending on the frequency) transducers spaced 1.2 m apart were deployed in the Connecticut River estuary both in the along stream and across stream direction to observe high Reynolds number stratified shear instabilities. In this presentation, results of these high-resolution temporal and spatial sampling measurements of shear instabilities are shown. These measurements demonstrate the utility of these techniques for improving our understanding of the evolution of shear instabilities.

3aUWb33. Is low frequency sound level uniformly increasing on a global scale? Jennifer L. Miksis-Olds (Appl. Res. Lab, Penn State, PO Box 30, Mailstop 3510D, State College, PA 16804, jlm91@psu.edu)

Deep water ambient sound level increases have been documented in the eastern North Pacific Ocean over the past 60 years. It remains unclear whether this increasing trend is observed in other regions of the world. In this work, data from the Comprehensive Nuclear Test Ban Treaty Organization International Monitoring System (CTBTO IMS) were used to examine the rate and direction of low frequency sound level change over the past decade in the Indian, South Atlantic, and Equatorial Pacific Oceans. The sources contributing to the overall sound level patterns differed between the regions. The dominant source observed in the South Atlantic was sound from seismic air gun surveys, while shipping and biologic sources contributed more to the acoustic environment at the Equatorial Pacific location. Unlike the increasing trend observed in the NE Pacific, sound levels over the past 5–6 years in the Equatorial Pacific were decreasing. Decreases were also observed for specific sound level parameters and frequency bands in the South Atlantic Ocean. Based on these observations, it does not appear that low frequency sound levels are increasing in all regions of the worlds' oceans. [Work supported by the Office of Naval Research.]

3aUWb34. Passive acoustics embedded on gliders—Weather observation through ambient noise. Pierre Cauchy, Pierre Testor, Laurent Mortier (LOCEAN, 4 Pl. jussieu, Paris 75252, France, pierre.cauchy@gmail.com), Laurent Beguery (DT INSU, Toulon, France), and Marie-Noelle Bouin (CMM, Brest, France)

Underwater gliders can provide high resolution water temperature and salinity profiles. Being able to associate them with a surface weather conditions estimation would allow to better study sea-air interactions. Since *in-situ* observations of the marine meteorological parameters are difficult, the development of a glider embedded weather sensor has been studied, based on the WOTAN approach. In the 1–30 kHz frequency range, the background underwater noise is dominated by wind generated noise. Focusing on the sound pressure level at 5, 8, 10, and 20 kHz allows to estimate the wind speed. Thus, deploying a glider with an embedded hydrophone gives an access to the surface weather conditions around its position. We have deployed gliders in the Mediterranean sea, with passive acoustic monitoring devices onboard. Four months of data have been recorded. Wind speed estimations have been confronted to weather buoys observations and atmospheric models predictions. Wind estimates have been obtained with a ~2 m/s error. A specific emphasis has been placed on the robustness of the processing through multi frequencies analysis and depth induced attenuation correction. A downscaling study has been performed on the acoustic sampling protocol, in order to meet the low energy consumption glider standards, for a future real time embedded processing. The glider generated noise and its vertical movement are not perturbing the estimation. Moreover, the surface behavior of the Slocum gliders allows an estimation of the wind direction.

3aUWb35. Hydroacoustic signals of Antarctic origin detected at ocean-bottom seismic stations off New Zealand. Justin S. Ball and Anne F. Sheehan (Geological Sci., CIRES/ Univ. of Colorado, 2200 Colorado Ave. #399, Boulder, CO 80309, justin.ball@colorado.edu)

Glacial calving from polar ice sheets is an important indicator of global climate change, and knowledge of ice discharge rates is useful for predicting global sea level variability and deep ocean circulation patterns. Since calving events are difficult to observe *in-situ* due to the remoteness of polar regions, the remote detection of hydroacoustic signals originating from these events is a useful monitoring tool for climatologists. We have observed hydroacoustic T-phases on an Ocean Bottom Seismic (OBS) network of 30 seismometers and differential pressure gauges that was deployed off the South Island of New Zealand in 2009–2010. These signals are emergent and strongly dispersive, containing most of their energy in the band between 5 and 15 Hz. We estimated backazimuths for events recorded on differential pressure gauges using array beamforming and F-K analysis. Our results suggest that some observed events originate in the vicinity of Ninnis and Mertz glaciers on George V Coast, East Antarctica, in general agreement with epicenters located by prior surface-wave based calving detection studies.

3aUWb36. Speculations on the cause of finite density and sound speed gradients near the interface of muddy sediments. Allan D. Pierce (Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY), and Joseph O. Fayton (Mathematical Sci., Rensselaer Polytechnic Inst., Portsmouth, Rhode Island)

Holland et al. [JASA (2013)] at the previous ASA meeting reported positive density gradients and negative sound speed gradients at the water-mud sediment interface and asked "what processes drive them." A derivation using the Mallock-Wood relations yields a simple high-porosity approximate relation giving a negative proportionality between sound speed gradient and density gradient independent of depth and porosity. It is also argued that the solid portion of the mud consists of tiny mineral platelets, each of which carries a net negative charge. The presence of dissolved salts causes each platelet to behave as an electrical quadrupole, so that there is, on the average, a small electrical repulsive force between the platelets. With gravity taken into account, the equilibrium separation distance between two parallel vertically aligned platelets is one to two orders of magnitude greater than a typical length scale of the face of a platelet. However, when platelets touch, edge to face, there is an attractive force between platelets, and the net effect is that the platelets tend to be separated at a much shorter distance than the stand-off distance deep within the sediment. Paper reports ongoing efforts to estimate the depth over which the transition occurs.

WEDNESDAY AFTERNOON, 7 MAY 2014 555 A/B, 1:30 P.M. TO 3:05 P.M.

Session 3pAAa

Architectural Acoustics: The Technical Committee on Architectural Acoustics Vern O. Knudsen Distinguished Lecture

David Lubman, Chair
dlacoustics, 14301 Middletown Ln., Westminster, CA 92683

Chair's Introduction—1:30

Invited Paper

1:35

3pAAa1. Is there any acoustical reason that supports non-rectangular concert hall design? Tapio Lokki (Media Technol., Aalto Univ., POBox 15500, Aalto 00076, Finland, Tapio.Lokki@aalto.fi)

The debate between shoe-box and non-shoe-box concert halls has been around for several decades. From the total concert experience point-of-view, there seems to be pros and cons for both designs. However, when only the acoustics of a hall are considered, the shoe-box design is very often preferred. This presentation discusses the basic differences between these two hall types, and how they affect, e.g., perceived bass, envelopment, clarity, intimacy, and loudness. The presentation is supported by recent data that was gathered in a large listening test during winter 2014. Typical sensory profiles of these two hall types are presented with links to listening test subjects' preferences. Moreover, the presentation will explain why shoe-box halls render larger dynamic range than other halls do. Even though room acoustics, defined with an impulse response, is linear, non-linear dynamic differences exist between halls due to the non-linear excitation (an orchestra) and non-linear human spatial hearing.

Session 3pAAb

Listening to the "Virtual Paul's Cross"—Auralizing 17th Century London II

Matthew Azevedo, Chair
Acentech Inc., 33 Moulton St., Cambridge, MA 02138

The purpose of this session is to provide an opportunity for people to listen to the Virtual Paul's Cross auralization, which allows listeners to experience John Donne's 1622 Gunpowder Day sermon while surrounded in three dimensions by a reactive crowd of up to five thousand, the bells of St. Paul's, and the ambient soundscape of 17th century London. The auralization allows for real-time changes to crowd size, listener position, the behavior of the intelligent agents which create the crowd reactions, and variations in the type and frequency of ambient sounds and requires over one hundred concurrent audio channels, a dozen channels of real-time convolution, hours of project-specific source recordings, and a complex network of intelligent and stochastic logical structures.

Session 3pAO

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

Timothy K. Stanton, Chair
Woods Hole Oceanogr. Inst., MS #11, Woods Hole, MA 02543-1053

Chair's Introduction—1:00

Invited Paper

1:05

3pAO1. Advances in remote inference of physical and biological parameters using acoustic scattering techniques: Mapping the ocean in broadband "color". Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu)

Active narrowband acoustic scattering techniques have been used for decades to infer the distribution of marine organisms, such as fish and zooplankton, and to image physical processes, such as bubbles, suspended sediments, internal waves, and microstructure. Accurately inferring relevant biological and physical parameters from the acoustic returns, such as size or abundance of organisms or intensity of mixing, has represented a far more formidable obstacle, requiring a multi-faceted approach in order to make significant headway. Over the years, advances have been made in understanding the fundamental scattering physics, resulting in more robust and accurate scattering models. These models have been guided and tested by controlled laboratory scattering experiments as well as in a plethora of field experiments. Rapid advances in instrumentation and deployment platforms have also enabled new insights to be gained. In this presentation, a brief overview of this research area is given, results from the development and implementation of broadband scattering techniques for studying physical and biological processes over relevant spatial and temporal scales are presented, and limitations of these techniques considered. Possible future directions and advances in the area of remote physical and biological parameter estimation from active acoustic scattering data will be discussed.

Session 3pBA

Biomedical Acoustics: Nonlinear Response of Encapsulated Microbubbles

Kausik Sarkar, Chair
Mech. and Aerosp. Eng., George Washington Univ., 801 22nd St. NW, Washington, DC 20052

Contributed Papers

1:00

3pBA1. Effects of ambient pressure variation on the subharmonic response from contrast microbubbles. NIma Mobadersany (George Washington Univ., Washington, Virginia), Amit Katiyar (Mech. Eng., Univ. of Delaware, Austin, Texas), and Kausik Sarkar (George Washington Univ., 801 22nd St. NW, Washington, DC 20052, sarkar@gwu.edu)

Ambient pressure dependent subharmonic response scattered from encapsulated contrast agent is investigated for non-invasive estimation of local blood pressure. Bubble dynamics is simulated using two different models of the encapsulation: the strain-softening elastic model and the Marmottant model. Unlike fundamental response, subharmonic response from a bubble occurs in a narrow range of excitation pressures—it appears only above a threshold excitation and disappears at higher excitation. The variation in subharmonic responses is chaotic with increasing ambient pressure at very low excitation frequencies. On the other hand, for high excitation frequencies, the subharmonic response increases with increasing ambient pressure For intermediate frequencies, the variation can be either monotonic increase, monotonic decrease, or nonmonotonic with increasing ambient pressure depending upon the excitation intensity. The simulated results will be discussed taking into account the effects of ambient pressure variation on the subharmonic threshold. [Partially supported by National Science Foundation.]

1:15

3pBA2. Ambient pressure estimation using subharmonic emissions from contrast microbubbles. Krishna N. Kumar, Shirshendu Paul (George Washington Univ., 801 22nd St. NW, Washington, VA 20052, krishnagwu@gwu.edu), and Kausik Sarkar (George Washington Univ., Washington, DC)

In cancerous tumors, the interstitial fluid pressure is higher than that in normal tissues, and therefore can be used as a diagnostic marker. Here we are presenting the results of an *in vitro* study aimed at developing an ultrasound-aided noninvasive pressure estimation technique using contrast agents—Definity®, a lipid coated microbubble, and an experimental poly lactic acid (PLA) microbubbles. Scattered responses from these bubbles have been measured *in vitro* as a function of ambient pressure using a 3.5 MHz acoustic excitation of varying amplitude. Definity bubbles produced stronger subharmonic than the PLA coated ones, and therefore, are better suited for this application. At an acoustic pressure of 500 kPa, Definity® microbubbles showed a linear decrease in subharmonic signal with increasing ambient pressure, registering a 12 dB reduction at an overpressure of 120 mm Hg. However, at other frequencies, the variation of subharmonic emission with ambient pressure is nonmonotomic as was also predicted by theoretical modeling in our group. [Partially supported by National Science Foundation.]

1:30

3pBA3. Acoustic characterization of polymer-encapsulated microbubbles with different shell-thickness-to-radius ratios using *in vitro* attenuation and scattering: Comparison between different rheological models. Lang Xia (Mech. and Aerosp. Eng., George Washington Univ., Washington, VA), Shirshendu Paul (Mech. and Aerosp. Eng., George Washington Univ., Washington, DC), Parag V. Chitnis, Jeffrey Ketterling (Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), Paul Sheeran, Paul Dayton (Biomedical Eng., Univ. of North Carolina Chapel Hill, Chapel Hill, NC), and Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., 801 22nd St. NW, Washington, DC 20052, sarkar@gwu.edu)

Acoustic behaviors of five different polymer (polylactide) encapsulated microbubbles—PB127 (Point Biomedical), PH37, PH43, PH44, PH45 (Phillips Healthcare) with different shell-thickness-to-radius ratios (STRR) of 3.5, 30, 40, 65, and 100 nm/μm have been characterized. *In vitro* attenuation data were used to determine the interfacial rheological properties of their shells. Use of different models—Newtonian, viscoelastic, strain-softening, Marmottant, and Church—resulted in similar rheological properties. The shell elasticity and shell viscosity were found to increase with increasing shell thickness as expected. The nonlinear scattered response from these microbubbles were measured. Experimentally measured scattered subharmonic response were compared with the model predictions.

1:45

3pBA4. Nonlinear intravascular ultrasound contrast imaging with a modified clinical system. Himanshu Shekhar, Ivy Awuor, Sahar Hashemgeloogerdi, and Marvin M. Doyley (Elec. and Comput. Eng., Univ. of Rochester, 212 Conant Rd. Apt. C, Rochester, NY 14623, himanshushekhar@rochester.edu)

An intravascular ultrasound system capable of visualizing microbubble contrast agents could provide functional information for assessing atherosclerotic plaques. The goal of this study was to investigate the feasibility of contrast-enhanced imaging with a modified commercial intravascular ultrasound system. We employed an iLab^{TM} system (Boston Scientific/Scimed, Natick, MA) equipped with an Atlantis^{TM} PV imaging catheter (15-MHz center frequency, 26% fractional bandwidth) to image tissue mimicking phantoms that had contrast agent (Targestar-P®, Targeson Inc., CA) flowing in side channels parallel to the center lumen. Chirp-coded pulses were employed with transmit frequency of 12 MHz and peak pressures ranging from 1–2 MPa. The ultraharmonic response (18 MHz) was isolated from the backscattered radio-frequency using pulse inversion and matched filtering, to produce contrast specific images. We evaluated the detection sensitivity of the agent as a function of microbubble concentration and transmit pulse parameters. The results revealed that side channels with diameters ranging from 500 μm to 2 mm could be visualized for a wide range of concentrations. These results demonstrate that functional imaging of plaque neovascularization is feasible with commercially available intravascular catheters. Further development of such systems can facilitate the widespread use of contrast-enhanced intravascular ultrasound for preclinical research and clinical imaging.

2:00

3pBA5. Microbubble spectroscopy of microbubble-loaded stem cells for targeted cell therapy. Tom Kokhuis, Ilya Skachkov (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Benno Naaijkens (Dept. of Pathol., VU Univ. MC, Amsterdam, Netherlands), Lynda Juffermans (Dept. of Physiol., VU Univ. MC, Amsterdam, Netherlands), Otto Kamp (Dept. of Cardiology, VU Univ. MC, Amsterdam, Netherlands), Ton van der Steen (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Michel Versluis (Phys. of Fluids group, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, m.versluis@utwente.nl), and Nico de Jong (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands)

Stem cells can be conjugated with targeted microbubbles to form highly echogenic complexes, dubbed StemBells. The complexes can improve stem cell delivery for the local repair of damaged cardiac tissue after a myocardial infarction through propulsion by acoustic radiation forces. While the first *in-vivo* tests hold great promise, the system would greatly benefit from a mapping of the acoustic parameter space. Here, we develop the theoretical background based on a modified Rayleigh-Plesset type equation to describe the dynamics of the StemBells in response to ultrasound. The complex is shown to resonate as a whole entity and resonance curves are constructed from numerical simulations resembling single bubble responses at a size that relates to the effective complex radius ~10 μm. Ultra high-speed optical imaging of single StemBell complexes at different frequencies using the microbubble spectroscopy method allows for a full characterization with excellent agreement with the developed model. Moreover, from the experimental resonance curves, we obtain values for the effective viscoelastic shell parameters of the StemBell complexes. These results have enabled the demonstration of the feasibility of manipulating StemBells inside chicken embryo microvasculature in an accompanying paper.

2:15

3pBA6. StemBells: Localized stem cell delivery using targeted microbubbles and acoustic radiation force. Tom Kokhuis, Ilya Skachkov (Biomedical Eng., Erasmus Medical Ctr., 's-Gravendijkwal 230, Faculty Bldg. (Rm. Ee2302), Rotterdam 3000 CA, Netherlands, t.kokhuis@erasmusmc.nl), Benno Naaijkens (Dept. of Pathol., VU Univ. Medical Ctr., Amsterdam, Netherlands), Lynda Juffermans (Dept. of Physiol., VU Univ. Medical Ctr., Amsterdam, Netherlands), Otto Kamp (Dept. of Cardiology, VU Univ. Medical Ctr., Amsterdam, Netherlands), Antonius van der Steen (Biomedical Eng., Erasmus Medical Ctr., Rotterdam, Netherlands), Michel Versluis (Phys. of Fluids Group, Univ. of Twente, Enschede, Netherlands), and Nico de Jong (Biomedical Eng., Erasmus Medical Ctr., Rotterdam, Netherlands)

The use of stem cells for regenerative tissue repair is hampered by the low number of cells delivered to the site of injury. To increase the delivery, we developed a new technique in which stem cells are coated with functionalized microbubbles, creating echogenic complexes dubbed StemBells. StemBells are highly susceptible to acoustic radiation force; this acoustic force can then be used after injection to deliver the StemBells locally at the treatment site. The dynamics of StemBells during ultrasound insonification was characterized using high-speed optical imaging and is described in an accompanying paper. Here, we investigate the feasibility of manipulating StemBells acoustically after injection employing a chicken embryo model, allowing for the real-time optical observation of the effects of acoustic radiation force *in vivo*. StemBells were infused by placing a custom-made catheter into one of the vitelline veins. Acoustic radiation force (1 MHz, P = 200–450 kPa, 10% duty cycle) was observed to propel StemBells from the centerline of the microvessels (200–500 μm) to the wall distal from the transducer. Peak translational velocities increased with pressure and varied between 50 μm/s to 300 μm/s. The acoustic radiation force had no effect on the trajectory of bare stem cells.

2:30

3pBA7. Frequency-sum passive cavitation imaging. Kevin J. Haworth, Kirthi Radhakrishnan (Internal Medicine, Univ. of Cincinnati, Biomedical Ultrason. and Cavitation Lab., Cincinnati, OH, kevin.haworth@uc.edu), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Passive cavitation imaging (PCI) is a method for spatially mapping acoustic emissions caused by microbubble activity, including subharmonic and ultraharmonic emissions that denote stable cavitation. The point spread function (PSF) of passive cavitation images is diffraction limited. When typical clinical diagnostic linear arrays are used for PCI, the diffraction limit results in high azimuthal resolution but low axial resolution. Abadi *et al.* (2013)recently demonstrated a method called frequency-sum beamforming, which employs second-order or higher products of the acoustic emissions to manufacture higher frequencies, thereby reducing the size of the PSF. We applied this approach to cavitation emissions recorded from albumin-shelled bubbles insonified by 2 MHz ultrasound. Cavitation emissions were recorded on a 5 MHz, 128 element linear array using a Vantage scanner (Verasonics Inc.). Quadratic and fourth-order frequency-sum beamforming was applied to both harmonic and ultraharmonic cavitation emissions. Corresponding simulations were also performed to illustrate frequency-sum passive cavitation imaging of multiple bubbles. In comparison to delay-and-sum PCI, apparent areas of cavitation activity decreased when products of the emissions were used to perform frequency-sum beamforming. However, frequency-sum beamforming also produced artifacts, including the appearance of spurious emission sources.

2:45

3pBA8. Estimation of damping coefficient based on the impulse response of echogenic liposomes. Jason L. Raymond (Dept. of Biomedical Eng., ThoraxCtr., Erasmus MC, CVC 3940, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, raymonjl@mail.uc.edu), Ying Luan, Tom van Rooij (Dept. of Biomedical Eng., ThoraxCtr., Erasmus MC, Rotterdam, Netherlands), Shao-Ling Huang, David D. McPherson (Dept. of Internal Medicine, Univ. of Texas Health Sci. Ctr., Houston, TX), Nico de Jong (Dept. of Biomedical Eng., ThoraxCtr., Erasmus MC, Netherlands, Netherlands), and Christy K. Holland (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Echogenic liposomes (ELIP) are under development as theragnostic ultrasound contrast agents for the diagnosis and treatment of cardiovascular disease. ELIP formulations have a phospholipid bilayer shell and are echogenic due to the presence of air; however, the exact location of the entrapped air has not been fully ascertained. Air pockets could either be stabilized by lipid monolayers within the liposome, or by the lipid bilayer shell. Our goal is to develop a more complete understanding of the encapsulation and shell properties of ELIP. This study demonstrates a method to estimate the damping coefficient using experimentally measured radius-time curves of the impulse response of individual ELIP using optical methods. The nondimensional damping coefficient and the natural frequency of oscillation were estimated based on 140 individual impulse responses as measured with the Brandaris 128 fast-framing camera (15 Mfps) at 37°C. The damping coefficient was in agreement with the damping coefficient as measured previously using a broadband acoustic technique [Raymond *et al.*, Ultrasound Med Biol. **40**(2), 410–421 (2014)]. However, the natural frequency of oscillation was lower than previously reported.

3:00

3pBA9. The stable nonlinear acoustic response of free-floating lipid-coated microbubbles. Ying Luan, Guillaume Renaud, Tom Kokhuis, Antonius van der Steen, and Nico de Jong (Biomedical Eng., Erasmus Medical Ctr., Pieter de Hoochweg 119A, Rotterdam 3024 BG, Netherlands, y.luan@erasmusmc.nl)

The stability of the microbubbles maintained by the lipid coating is crucial for diagnostic contrast-enhanced ultrasound imaging. We present a study of the stability of the dynamic response of single free-floating microbubbles (DSPC-based homemade microbubbles) with an acoustical camera. A 30 MHz probing wave (800 μs duration) measures the dynamic response of single microbubbles to 42 short sine bursts (1 MHz, 10 μs duration, 3 μs interval between each excitation) at three different peak pressures (25, 100,

and 200 kPa). For each microbubble exposed to the 42 consecutive pulses, the following parameters were calculated: the radial strain at the driving frequency (ε_f), at the second/third harmonic frequencies (ε_{2f},ε_{3f}), the ratio of radial excursion in expansion over that in compression (EoC) and the dc offset in the time-domain response. Nearly all 1500 individual microbubbles measured showed stable vibrational response. As expected ε_{2f} and ε_{3f} increase with ε_f, but they reach a plateau when ε_f exceeds about 30%. For ε_f smaller than 15%, we observed compression-dominant behaviors (dc offset < 0 and EoC < 1), while microbubbles show expansion-dominant responses (dc offset > 0 and EoC > 1) for ε_f larger than 15%.

WEDNESDAY AFTERNOON, 7 MAY 2014 — 557, 1:30 P.M. TO 2:50 P.M.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Tessa Bent, Chair
Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405

Chair's Introduction—1:30

Invited Papers

1:35

3pID1. Hot Topics—Hidden hearing loss: Permanent cochlear-nerve degeneration after temporary noise-induced threshold shift. M. Charles Liberman and Sharon G. Kujawa (Eaton Peabody Labs., Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA 02114, charles_liberman@meei.harvard.edu)

The classic view of sensorineural hearing loss (SNHL) is that the "primary" targets are hair cells, and that cochlear-nerve loss is "secondary" to hair cell degeneration. Our work in mouse and guinea pig has challenged that view. In noise-induced hearing loss, exposures causing only reversible threshold shifts (and no hair cell loss) nevertheless cause permanent loss of >50% of cochlear-nerve/hair-cell synapses. Similarly, in age-related hearing loss, degeneration of cochlear synapses precedes both hair cell loss and threshold elevation. This primary neural degeneration has remained hidden for two reasons: (1) the spiral ganglion cells, the cochlear neural elements commonly assessed in studies of SNHL, survive for years despite loss of synaptic connection with hair cells, and (2) the degeneration is selective for cochlear-nerve fibers with high thresholds. Although not required for threshold detection in quiet (e.g., threshold audiometry, auditory brainstem response threshold), these high-threshold fibers are critical for hearing in noisy environments. Our research suggests that (1) primary neural degeneration is an important contributor to the perceptual handicap in SNHL, and (2) noise exposure guidelines should be re-evaluated, as they are based on the faulty premise that threshold audiograms are the most sensitive measures of noise damage.

2:00

3pID2. Energy harvesting from acoustic fields. Kenneth Cunefare (Georgia Tech, Mech. Eng., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

While energy harvesting from a variety of ambient sources (vibration, light, and wind) has been demonstrated and sensing and communication applications to exploit those sources have been developed, acoustic energy as an ambient energy source has not received similar attention, except for a few very specialized special cases; this "hot topic" will focus on the special cases. The reason for otherwise limited development comes down to the basic physics of how much energy is available within a "typical" acoustic field. For airborne sounds, the energy density in sound fields that are perceived by humans to be loud to painfully loud (e.g., 80 to 140 dB, or ~0.2 Pa to ~200 Pa) actually represent an extremely low available energy source. In consequence, means must be taken to intensify an acoustic response, for example through resonance, but even so, the available energy remains limited. But, what if the sound field is not "typical"? One of the exceptions which enables viable acoustic energy harvesting is the sound field that exists inside of an operating jet aircraft engine. Another exception is within pumped and pressurized fluid systems, where acoustic pressure variations may reach into the mega-Pascal (MPa) range. Energy harvesting from such a fluid-borne acoustic source is feasible for powering sensors and wireless communication systems, has been successfully demonstrated, and may yield commercialized technology within only a few years.

2:25

3pID3. Underwater sound from pile driving, what is it and why does it matter. Peter H. Dahl, Per G. Reinhall (Appl. Phys. Lab. and Dept. of Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu), Arthur N. Popper (Dept. of Biology, Univ. of Maryland, College Park, MD), Mardi C. Hastings (Dept. of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), and Michael A. Ainslie (Acoust. and Sonar Group, TNO, The Hague, Netherlands)

Pile driving as used for in-water construction can produce high levels of underwater sound that has potential to produce physiological and/or behavioral effects on fish, benthic invertebrates, and marine mammals. There are two basic pile driving methods: impact pile driving where the pile is driven by strikes from a high-energy hammer, and vibratory pile driving where the pile is effectively vibrated into the sediment. Often both methods are used on the same pile. At ranges on the order of 10 m, and considering steel piles of diameter 0.75–1 m, vibratory pile driving produces underwater sound pressures of order 100–1000 Pa, which is often sustained for minutes. In contrast, each impact pile strike produces peak sound pressures on the order of 100 kPa, with effective duration of the sound from the strike being of order tens of milliseconds. Measurements made both far from the pile source (range many depths) and close-in (range of about 1–2 depths) for impact and vibratory pile driving are presented, along with examples modeling of such sound. We conclude by showing why such sounds matter to aquatic life; potential effects include injury at close range and behavioral changes, including evasion resulting in habitat loss at greater distance.

WEDNESDAY AFTERNOON, 7 MAY 2014 551 A/B, 12:55 P.M. TO 2:45 P.M.

Session 3pPA

Physical Acoustics: Topics in Nonlinear Acoustics

Dimitri Donskoy, Chair
Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030

Chair's Introduction—12:55

Contributed Papers

1:00

3pPA1. The phenomenon of self-trapping of a strongly nonlinear wave. Oleg Rudenko (Acoust., Moscow and Nizhni Novgorod Univ., General Phys. and Earth Phys. Inst. of RAS, Moscow State Univ., Moscow, Russian Federation, rudenko@acs366.phys.msu.ru) and Claes Hedberg (Eng. Sci., Blekinge Inst. of Technol., Karlskrona, Sweden)

Self means here an effect of a wave on itself. Several self-action phenomena are known in nonlinear wave physics. Among them are self-focusing of beams, self-compression of light pulses, self-channeling, self-reflection (or self-splitting) waves with shock fronts, self-induced transparency, and self-modulation. These phenomena are known for weakly nonlinear waves of different physical origin. Our presentation at ASA meeting in Montreal [POMA **19**, 045080 (2013)] was devoted to strongly nonlinear waves having no transition to the linear limit at infinitesimally small amplitudes. Such waves can demonstrate particle-like properties. Self-trapping consists of the arrest of wave propagation and in the formation of a localized state. In particular, the model generalizing the Heisenberg' ordinary differential equation to spatially distributed systems predicts periodic oscillations but no traveling waves. Different models for strongly nonlinear waves will be considered and some unusual phenomena will be discussed. Preliminary results were published in Ac. Phys. **59**, 584 (2013) and Physics-Uspekhi (Adv. Phys. Sci.) **183**, 683 (2013). [This work was supported by the Megagrant No.11.G34.31.066 (Russia) and the KK Foundation (Sweden).]

1:15

3pPA2. Nonlinear acoustic waves in media with hysteresis and long-time relaxation. Lev A. Ostrovsky (PSD, NOAA ESRL, 325 Broadway, R/PSD99, Boulder, CO 80305, lev.a.ostrovsky@noaa.gov)

It is known that the media with complex structures (e.g., rocks and ceramics) possess an anomalously strong elastic nonlinearity. Nonlinear acoustic effects are used in a number of important applications, such as seismic waves (earthquakes), non-destructive testing, and contact physics. Two specific features were registered, in most of the experiments: hysteresis in the stress-strain relation and longtime relaxation (slow time). Some physical models, mostly phenomenological, have been suggested to explain these phenomena. However, there are very few works considering the effect of medium hysteresis and relaxation on wave propagation and oscillations in resonators. This presentation is a review of both published and new results in this area. Among the problems discussed are: (1) A brief overview of experiments and models; (2) Analysis of wave propagation in simplified models of media with hysteresis as opposed to the nonlinear waves in media characterized by elastic constants of second and third order; (3) Theory of wave interaction in resonators with hysteresis; (4) Study of waves in media with slow time relaxation; (5) Some data from laboratory and field experiments; (6) Some unsolved issues and the relevant future work will be outlined in conclusion.

1:30

3pPA3. Development of high intensity focused ultrasound transducers to deliver specified shock wave amplitudes at the focus. Vera A. Khokhlova, Petr V. Yuldashev, Pavel B. Rosnitskiy (Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow, Russian Federation, vera@apl.washington.edu), Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), and Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

High intensity focused ultrasound (HIFU) is currently emerging into many clinical applications. Certain HIFU modalities, histotripsy, for example, rely on the formation of high amplitude shocks in the focal waveform of the beam. These shocks develop at different focal pressure levels depending on the geometry of the HIFU transducer. The goal here was to determine

optimal transducer parameters that would result in specified shock amplitudes and corresponding peak negative focal pressures. The hypothesis was that pressure level for shocks to form is mainly determined by the F-number of the transducer. As nonlinear effects accumulate almost entirely in the focal lobe of HIFU beams, shocks will form at the higher pressures for lower F-number transducers with shorter focal lobe and thus will have higher amplitudes. Simulations based on the Khokhlov-Zabolotskaya-Kuznetsov equation have shown that for typical HIFU transducers with 1–3 MHz frequencies, geometries with F-number close to 1 are optimal for generating waveforms with about 70 MPa shocks and 12 MPa peak negative pressures. For lower F-number transducers, higher amplitude shocks and peak negative pressures will be formed unless cavitation occurs proximal to the focus to attenuate the focal beam. [Work supported by the grants MK-5895.2013.2, RFBR-13-02-0018, NIH-EB007643, and T32-DK007779.]

1:45

3pPA4. Nonlinear collapse of vortex. Konstantin A. Naugolnykh (Phys., Environ. Study Res. Lab., NOAA, 325 Broadway, Boulder, CO 80305, konstantin.naugolnykh@noaa.gov)

Sound radiation by two vortices with different intensity and sign changes the distribution of vortex field. The acoustic instability and vortices motion may occur in such a system. The linear stage of this process was considered previously the spread of similar vortices (Klyatskin, 1966) and attraction of different vortices (Gryanik, 1983; Kop'ev and leont'ev, 1983). As a result of attraction, the nonlinear effect of collapse of vortices may appear. This process is considered by the method of matched asymptotic expansion of the solution for the point vortices in an incompressible fluid and the solution of nonlinear Burgers equation. The features of cylindrical wave spread and nonlinear distortion both indicated.

2:00

3pPA5. Enhancement of a general solution to Lighthill-Westervelt nonlinear acoustic equation to the cases of inhomogeneous and random media. Harvey C. Woodsum (Hobbit Wave, Inc., 21 Continental Blvd., Merrimack, NH 03054, hwhwoodsum@gmail.com)

A general solution to the Lighthill-Westervelt equation of nonlinear acoustics, previously developed and successfully applied to model the scattering of sound by sound and to the parametric array, has been generalized further for use in the cases of inhomogeneous and random media. The form of the solution makes use of an exact inverse differential operator in combination with a sequence of perturbation terms that comprise the multiple orders of nonlinear acoustic scattering to arbitrary order. Integration techniques have been developed which allow accurate, approximate, analytical solutions under particular circumstances. These solutions are shown to reduce to other previously known solutions in the appropriate limits.

2:15

3pPA6. Analysis of scattering from dual-frequency Incident beam interaction. Chrisna Nguon (Univ. of Massachusetts Lowell, 63 Hemlock St., Dracut, MA 01826, chrisna_Nguon@student.uml.edu), Nicholas Misiunas, Barbara Deschamp, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, Lowell, MA)

The characterization of acoustically inhomogeneous structures insonified by dual-frequency acoustic beams is undertaken. A layer of microbubble contrast agents is included to improve the resolution of the scattered field. Of particular interest is the identification of characteristic sizes of volume scatterers and their principal orientations by measurement of their acoustic signatures in the exterior field. The inclusion of bubble contrast agents has shown to enhance the directional pattern of the back-scattered field generated from the difference frequency source created by the nonlinear interaction of the incident beams. However, the sensitivity of the scattered field to variations in the contrast parameter and spatial orientation requires computation to be carried out in a high-dimensional parameter space that includes the difference wavenumber, angle of incidence, acoustic nonlinearity parameter, and several microbubble parameters. A fast computation of the exterior field is carried out using a monopole expansion of the Green's function that separates the source and observation coordinates. The exterior field is sampled and analyzed in the aforementioned system parameter space to identify the weighted combination of parameters that serve as a minimum representation for detecting the changes in the medium parameters through measurements of the exterior scattered field.

2:30

3pPA7. Analysis of strongly nonlinear blast waves using the Rankine-Hugoniot relations and the nonlinear ray theory. Jae-Wan Lee, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., Yonsei University, 50 Yonsei-ro, Seodaemun-gu, Seoul 120749, South Korea, fantalex@yonsei.ac.kr), and Woosup Shim (The 5th R&D Institue-2, Agency for Defense Development, Daejeon, Daejeon, South Korea)

Blast waves produced by such events as nuclear explosions can be considered as strongly-nonlinear shock waves, the description of which requires a theoretical framework more accurate than the second-order wave equation. In this regard, computational fluid dynamics (CFD) techniques based on the Euler equation are frequently used. However, CFD techniques are very time-consuming for blast waves traveling over great distances. This paper presents a theoretical framework for propagation of strongly nonlinear blast waves, which shines in both speed and accuracy. Local propagation speeds of a blast wave are obtained by applying the Rankine-Hugoniot relations to "infinitesimal shocks" between adjacent phase points comprising the blast wave. By grafting the propagation speed onto the nonlinear ray theory, the evolution of the blast wave can be computed. Phenomena characteristic of strongly-nonlinear blast waves such as the Mach reflection, the Mach stem generation, and the self-refraction are observed from numerical simulations.

Session 3pSA

Structural Acoustics and Vibration, Noise, and Architectural Acoustics: Acoustics of Sports

Matthew D. Shaw, Cochair
Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802

Donald B. Bliss, Cochair
Mech. Eng., Duke Univ., 148B Hudson Hall, Durham, NC 27705

Chair's Introduction—1:25

Invited Papers

1:30

3pSA1. Vibrational assessment of wood, composite, and plastic hurleys. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu)

The Gaelic sport of hurling combines elements of field and ice hockey, lacrosse, and even baseball. The hurling stick, or hurley, has a long narrow handle which tapers to a large flat paddle, or bas, and is primarily made from white ash. Sticks show variation in the thickness, size, and shape of the paddle and composite hurleys have recently been introduced to the game. In this paper, we use experimental modal analysis to study the vibrational mode shapes and frequencies of several ash and composite hurleys, in adult and youth sizes, including the infamous 1970's Wavin plastic hurley which was quickly abandoned due to excessive vibration and sting. Bending and torsional mode shapes are found to be similar to those in baseball bats and field hockey sticks. A third family of vibrational modes exhibiting bending in the handle and torsion in the paddle are similar to vibrational modes observed in ice and field hockey sticks. These three types of mode shapes help define the sweet spot as well as influencing the perception of feel in the hands of a player.

1:50

3pSA2. Remotely monitoring performance in sports acoustically. David Browning (139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com) and Peter Scheifele (Dept. of Comm. Sci., Univ. of Cincinnati, Cincinnati, OH)

In many sports, a coach will tell you that their trained ear can detect a superior performance. Volleyballs, soccerballs, and footballs all resonate at a characteristic frequency when struck. The resulting sound can be monitored off the field of play to determine how hard they were hit. A simple sound level meter can be easily modified to make a smackmeter. A bit more complicated but perhaps more rewarding is the resonate sound from aluminum baseball bats, or tennis and squash reckets, when can reveal not only how well but where the hit was made. Monitoring the sound of the stumming shaft of a golf club during a swing gives valuable information on swing speed and uniformity. Most every sport, for example, how about the splash of a dive, appears to have sounds that could be worthwhile to remotely monitor, especially given the resolution and speed of modern analysis techniques.

2:10

3pSA3. Racquetball exposed: Analysis of decay times and exposure levels in racquetball courts. Matthew D. Shaw and Eric C. Mitchell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mdshaw16@gmail.com)

Racquetball courts are interesting acoustic spaces. The architecture varies from court to court; therefore, the acoustic environment also changes. This talk will present reverberation time measurements of two types of courts: fully enclosed courts and courts with open viewing areas. Similarities and differences in the decay times will be discussed. This research will also investigate a player's exposure level during a typical racquetball match.

Contributed Papers

2:30

3pSA4. Sound transmission issues in higher-story fitness facilities. Sharon Paley (ECORE Int., 715 Fountain Ave., Lancaster, PA 17601, sharon.paley@ecoreintl.com)

The growing popularity of Crossfit and other high-intensity workouts coupled with an increasing trend for gyms/weight rooms to be installed on higher stories of lighter-weight buildings is casting a new light on sound transmission issues in fitness facilities for the acoustic consultant. Traditional gym flooring and mats do not sufficiently minimize the impact of a dropped dumbbell for the tenant below. This is a growing problem for both residential gyms found in multi-family housing as well as the more extreme facilities for professional athletes and olympians. Field test results, limitations of existing field measurement techniques, and potential solutions will be discussed.

2:45

3pSA5. Sound and noise in high schools gymnasiums. Antonio P. Carvalho and Carla C. Barreira (Lab. of Acoust., College of Eng. (FEUP), Univ. of Porto, FEUP (NIF501413197), R. Dr. Roberto Frias, Porto 4200-465, Portugal, carvalho@fe.up.pt)

The goal of this research was to characterize the interior acoustics of high schools sports facilities using objective parameters. In situ measurements were done in 68 school gymnasiums in Portugal (volume from 450 to 2680 m^3) regarding LAeqBN (background noise without gym classes), LAeqPE (ongoing Physical Education classes), RT, and RASTI. The results for LAeqBN were from 34 dB (L90) to 50 dB (L10) with a median of 42 dB. For the LAeqPE were found values from 75 dB (L90) to 85 dB (L10) with a median of 80 dB. For the RT(500/1k/2k) room values from 2.6 s (RT_{90}%) to 6.9 s (RT_{10}%) with a median of 4.8 s, were measured. The room average RASTI values were from 0.27 ($RASTI_{90}$%) to 0.43 ($RASTI_{10}$%) with a median of 0.34. These sports rooms proved to be highly reverberant, almost without sound absorbing materials, which might be harmful, especially for the gym teachers. The subjective perception of the PE teachers was analyzed through questionnaires where it was verified that they feel most discomfort when it comes for noise (and thermal) conditions. This was supported by the objective results obtained. Ideal values for those acoustic parameters are presented.

WEDNESDAY AFTERNOON, 7 MAY 2014 BALLROOM D, 1:00 P.M. TO 3:00 P.M.

Session 3pSC

Speech Communication: Developmental Topics in Speech Communication

Linda Polka, Chair

School of Commun. Sci. and Disorder, McGill Univ., 1266 Pine Ave., West, Montreal, QC H3G 1A8, Canada

Contributed Papers

1:00

3pSC1. Pre-babbling infants prefer listening to infant speech: Implications for vocal learning in humans. Matthew Masapollo, Linda Polka (McGill Univ., 1266 Pine Ave. West, Montreal, QC H3G 1A8, Canada, matthew.masapollo@mail.mcgill.ca), and Lucie Ménard (Univ. of PQ at Montreal, Montreal, QC, Canada)

For human infants to engage in vocal learning, they must effectively monitor and assess their own self-produced speech, which entails perceiving speech produced by an infant. Yet, little is known about how infants respond to infant-produced speech. Here, we demonstrate that pre-babbling infants prefer listening to infant speech. Across four experiments, 3-to-6-month-olds were tested in a preferential listening procedure, using vowels synthesized to emulate productions by female adults and infants. In experiment 1, infants listened longer to vowels produced by infant than adult speakers. However, in experiment 2, infants failed to show any listening preference for infant versus adult vowels synthesized with matching, infant-appropriate pitch values, suggesting that infants were either attracted to higher voice pitch *per se* or to infant-like voice pitch. Failing to support a bias of the first type, infants in experiment 3 showed no listening preference when presented infant vowels with higher and lower infant-appropriate pitch values. Moreover, in experiment 4, infants showed a preference for infant versus adult vowels when synthesized with pitch values that are appropriate for a female using adult-directed speech; this suggests that infants are also attracted to infant vocal resonance properties. The implications of these results for speech development are discussed.

1:15

3pSC2. Relating moms Apostrophe productions of infant directed speech with their babies Apostrophe ability to discriminate speech: A brain measure study with monolingual and bilingual infants. Adrian Garcia-Sierra (I-LABS, Univ. of Washington, 850 Bolton Rd., Unit 1085, Storrs, Connecticut 06269, adrian.garcia-sierra@uconn.edu), Nairan Ramirez-Esparza (Psych. & Speech, Lang. and Hearing Sci., Univ. of Connecticut, Storrs, CT), Melanie S. Fish, and Patricia K. Kuhl (I-LABS, Univ. of Washington, Seattle, WA)

We report the benefit of Infant Directed Speech (IDS) on speech discrimination in 11- and 14 month-old monolingual (N = 16) and bilingual infants (N = 14). Mothers were instructed to read a booklet to their infants that contained sentences with target words (e.g., park, tear, bark, deer, etc.) at least once a day for four days. This activity was recorded by the LENA digital recorder that the infants were carrying around as they went about their lives. IDS was defined as the mothers' voice-onset time (VOT) durations when reading the booklet at home to their infants. Adult Directed Speech Mothers' IDS productions were correlated with their infants' brain responses associated with speech discrimination (Mismatch Negativity Response or MMR). The results showed that mothers' IDS correlated positively with the bilingual infants' positive-MMR and the monolingual infants' negative-MMR. Bilinguals' positive-MMR is interpreted as a less mature brain response since the positivity declines with age and a negative-MMR (adult-like) emerges later in development. These results show that even though the monolinguals and the bilinguals are at different developmental stages—as demonstrated by their MMRs— both monolingual and bilingual infants benefit from IDS to learn the sounds of their native language(s).

1:30

3pSC3. Infant-directed speech reduces English-learning infants preference for strong/weak versus weak/strong words. Derek Houston and Crystal Spann (Otolaryngol. - Head & Neck Surgery, Indiana Univ. School of Medicine, 699 Riley Hospital Dr., RR044, Indianapolis, IN 46202, dmhousto@indiana.edu)

Mounting evidence suggests that infant-directed speech (IDS) facilitates aspects of language acquisition. An important milestone of English-learning infants' language acquisition is encoding the predominant stress-initial rhythmic structure of English, which 9-month-olds have demonstrated by showing a looking-time preference for lists of strong/weak words (e.g., doctor) versus weak/strong words (e.g., guitar) (Jusczyk *et al.*, 1993). We tested for this preference in 48 9-month-olds using the headturn preference procedure. Twenty-four infants were presented with the words using IDS, and 24 were presented with adult-directed speech (ADS). Infants were presented with a visual display of a blinking light on one of two (left and right) monitors. When they oriented toward the monitor, a list was presented from behind that monitor until they looked away for more than 2 s. They were presented with lists of weak/strong words for half the trials and strong/weak words for the other half in quasi-random order. A repeated-measures AVOVA revealed a statistically significant interaction between word-type preference and speech condition. To our surprise, only infants in the ADS condition showed a preference strong/weak words. The findings raise the possibility that rhythmic properties of words may be more difficult for infants to encode in IDS than ADS.

1:45

3pSC4. Differences in the acoustic correlates of intonation in child and adult speech. Jill C. Thorson and James L. Morgan (Dept. of Cognit., Linguistic and Psychol. Sci., Brown Univ., 190 Thayer St., Box 1821, Providence, RI 02912, jill_thorson@brown.edu)

During speech perception, toddlers use prosodic cues, such as pitch, in order to identify the most salient/prominent information in the discourse. This process is critical in facilitating early word recognition and learning. How do children then use these acoustic cues in their own speech in comparison to mature adult speech? The motivation for this study is to examine the acoustic correlates of intonation employed by child (mean: 2.5 years) and adult speakers of English during a guided spontaneous production task. During an interactive game, we elicit a set of target nouns and label them as one of three types: (1) new, uttered for the first time by the participant, (2) given, previously uttered at least once by the participant, or (3) contrastive, uttered in direct opposition to a previously mentioned referent. Along with labeling the pitch accent, we measured f0 range, f0 slope, duration, and intensity for each target word under these varying conditions. These measurements allow us to compare the types of pitch accents used by children and adults, and how each group employs the acoustic correlates of intonation. Identifying how these parameters are used during production is an important step in understanding speech development in early language acquisition.

2:00

3pSC5. Mothers do not enhance phonemic contrasts of Mandarin lexical tones in child-directed speech. Puisan Wong (Speech and Hearing Sci., The Univ. of Hong Kong, 7/F, Meng Wah Complex, Faculty of Education, Pokfulam, Hong Kong, pswResearch@gmail.com), Xin Wang, Wenna Xi, Lingzhi Li, and Xin Yu (The Ohio State Univ., Columbus, OH)

Child-directed speech is characterized by higher pitch, more expanded pitch contours, and more exaggerated phonetic contrasts, which was suggested to facilitate speech sound acquisition. This study examined the perceptual and acoustic differences of mothers' disyllabic Mandarin lexical tones directed to adults and children to determine whether mothers exaggerated the pitch targets of the four Mandarin tones when speaking to children. Twelve Mandarin-speaking mothers produced 700 child-directed (CD) and adult-directed (AD) disyllabic words in a picture naming task. Five Mandarin-native speakers identified the mothers' AD and CD tones in filtered speech. Overall, CD lexical tones were identified with significantly lower accuracy than AD lexical tones (89% vs. 94%, $S = 25.5$, $p = 0.006$, $r = 0.927$, Wilcoxon Signed Rank Test). Acoustic analysis showed that the mean fundamental frequency (f0) of the four tones in both syllables was significantly higher in CD than in AD productions. No difference was found between AD and CD in the distinctive pitch targets for the 4 tones, namely pitch shift for Tone1, F0 slope for Tone2, minimum F0 for Tone3 and F0 slope for Tone4. F0 plots showed mostly parallel contours in AD and CD productions without exaggeration of the phonetic contrasts of the tones. [Work supported by NIDCD F31 DC008470-01A1.]

2:15

3pSC6. Perceptual acuity and production distinctness in child speech: Data from American English /r/. Tara McAllister Byun (Communicative Sci. and Disord., New York Univ., 665 Broadway, New York, NY 10012, tara.byun@nyu.edu) and Mark Tiede (Haskins Labs., New Haven, CT)

Previous work has shown that some adult listeners have more sharply defined perceptual categories than others, and listeners who have the most precise auditory targets also tend to produce more robust contrasts (e.g., Newman, 2003; Perkell *et al.*, 2004; Ghosh *et al.*, 2010). While it is likely that this relationship also holds in child/adolescent speakers, the hypothesis has not been directly tested. This study compared perception and production of the English /r-w/ contrast in 20 typically developing children aged 9–14. Two 10-step *rake-wake* continua were synthesized using *rake* tokens elicited from two child speakers (8-year-old male, 10-year-old female). Items were presented 8 times each in random order in a forced-choice identification task. Participants also produced *rake* and *wake* in a carrier phrase in casual and careful speech. Perceptual sensitivity was evaluated with a logistic function fitted over the number of *wake* responses at each step in the continuum. Preliminary results show considerable across-participant variation in the slope of the best-fit logistic function, the location of the perceptual boundary, and the acoustic distinctness of /r-w/ in production. We expect to demonstrate a relationship between perceptual acuity and production distinctness similar to that seen in adults. [Work supported by NIH.]

2:30

3pSC7. Timecourse of word recognition for listener-oriented speaking style adaptations. Suzanne V. van der Feest (Commun. Sci. and Disord., Univ. of Texas at Austin, 2504 Whitis Ave. A1100, Austin, TX 78712, suzanne@austin.utexas.edu) and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX)

Previous research has found that listener-oriented speaking style adaptations, such as Infant Directed Speech (IDS) and Clear Speech (CS), aid perception and development and improve intelligibility for children and adults, respectively. However, less is known about whether these different speaking styles enhance intelligibility in general, or if young children benefit most from IDS features aimed at capturing and maintaining attention. Additionally, information about time course differences in word recognition for these speech modifications has not been investigated. This study investigates both questions by presenting young adult listeners with semantically anomalous and meaningful sentences produced in IDS, CS, and Conversational (CO) Speech. Listeners participated in (1) a word-recognition-in-noise test and (2) a visual word recognition paradigm tracking eye movements. Results showed facilitated word recognition in noise for both IDS and CS compared to CO. Both speaking style adjustments also resulted in increased speed of word recognition in the visual paradigm. Semantic context provided additional facilitation only when more exaggerated acoustic cues were present (in IDS and CS). Preliminary analyses show that even for adult listeners IDS provided the greatest perceptual benefits, indicating that established advantages of IDS for young listeners cannot be mainly attributed to affect in addition to enhanced acoustic cues.

2:45–3:00 Panel Discussion

Plenary Session and Awards Ceremony

James H. Miller
President, Acoustical Society of America

Presentation of Certificates to New Fellows

Caroline Abdala - For contributions to understanding the postnatal maturation of the human cochlea, auditory nervous system, and middle ear

Judit Angster - For contributions to the acoustics of the pipe organ

David A. Brown - For contributions to fiber-optic and piezoelectric transduction science, and leadership in acoustics education

John R. Buck - For contributions to applications of random matrix and information theory in acoustic signal processing and bioacoustics

John A. Colosi - For contributions to the science of wave propagation in random media

Huangpin Dai - For contributions to the theory and methodology of the study of central processes in auditory perception

Michael J. Epstein - For integration of physiological and psychological processing in the perception of loudness

Vitalyi E. Gusev - For contributions to nonlinear acoustics and laser-based ultrasonic pulse generation

David R. Schwind - For contributions to the acoustical design of theaters, concert halls, and film studios

Bridget M. Shield - For research, teaching, and leadership to standardize classroom acoustics

Preston S. Wilson - For contributions to the theory and applications of the acoustics of bubbly media

Joseph N. Soker - For contributions to acoustical design and noise control applications in buildings and communities

Presentation of Awards

William and Christine Hartmann Prize in Auditory Neuroscience to Egbert de Boer

Medwin Prize in Acoustical Oceanography to Andone C. Lavery

R. Bruce Lindsay Award to Matthew J. Goupell

Helmholtz-Rayleigh Interdisciplinary Silver Medal to Mark F. Hamilton

Gold Medal to Brian C. J. Moore

Vice-President Gavel to Peter H. Dahl

President’s Tuning fork to James H. Miller

Session 3eED

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

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

Cameron T. Vongsawad, Cochair
Phys. & Astronomy, Brigham Young Univ., 1041 E. Briar Ave., Provo, UT 84604

This workshop for Providence area Girl Scouts (ages 12-17) consists of a hands-on tutorial, interactive demonstrations, and a panel discussion about careers in acoustics. The primary goals of this workshop are to expose the girls to opportunities in science and engineering and to interact with professionals in many areas of acoustics. A large number of volunteers are needed to make this a success. Please email Traci Neilsen (tnb@byu.edu) if you have time to help with either guiding the girls to the event and helping them get started (5:00 p.m. - 6:00 p.m.) or exploring principles and applications of acoustics with small groups of girls (5:00 p.m. - 7:30 p.m.). We will provide many demonstrations, but feel free to contact us if you would like to bring your own.

WEDNESDAY EVENING, 7 MAY 2014 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings.

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 Wednesday are as follows:

7:30 p.m.	Signal Processing in Acoustics	555AB