Session 4aAA

Architectural Acoustics: Assorted Topics in Architectural Acoustics I

Shinsuke Nakanishi, Cochair
Department of Engineering, Hiroshima International University, 5-1-1, Hiro-koshingai, Kure 737-0112, Japan

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

Contributed Papers

8:30

4aAA1. Design, construction, and evaluation of an omni-directional loudspeaker (Dodecahedron). Maryam Landi, Vahid Naderyan (Dept. of Phys. and Astronomy & National Ctr. for Phys. Acoust., Univ. of MS, NCPA, 145 Hill Dr., University, MS 38677, mlandi@go.olemiss.edu), and David S. Woolworth (Roland, Woolworth & Assoc., Oxford, MS)

Dodecahedron loudspeaker (Dodec) is an omni-directional sound source in the shape of a 12-sided loudspeaker with each side being a pentagon. The omni-directionality of this sound source makes it mainly applicable in room acoustical and sound insulation measurements and research as it can excite and saturate the room as much as possible. Dodec is a good approximation of a point sound source. Commercially available omni-directional loudspeakers are not economically efficient for some purposes. This paper outlines a simple and inexpensive Dodec loudspeaker designed, constructed, and evaluated using ISO 140 and ISO 3382 standards as reference as part of an independent coursework. The Dodec was evaluated in an anechoic chamber. The measured directivity of this Dodec meets the standards for omnidirectionality at all frequencies.

8:45

4aAA2. Room acoustic measurement techniques for large venues (greater than 10,000 cubic meters) in 2016. Jason R. Duty, Jeremy L. Decker, and Peter K. Holst (Charles Salter Assoc., 130 Sutter St., Fl. 5, San Francisco, CA 94104, jason.duty@cmsalter.com)

From sine sweeps to maximum-length sequence (MLS) type signals, balloon explosions to the yachting cannon, options to measure room acoustics in large venues have increased over the past couple of decades. There are new developments by academia and equipment manufacturers, as well as improvements on older methods. What are these new methods? What acoustical characteristics can one measure with each method? This presentation describes the different source options to measure room acoustical characteristics in larger spaces.

9:00

4aAA3. Acoustical energy relations in auditoriums. Jianliang Gao, Shiu-Keung Tang (Bldg. Services Eng., The Hong Kong Polytechnic Univ., Rm. ZS801, 8/F, Block Z, 181 Chatham Rd. South, Hunghom, Kowloon 999077, Hong Kong, jianliang.gao@connect.polyu.hk), and Yuezhe Zhao (State Key Lab. of Subtropical Bldg. Sci., South China Univ. of Technol., Guangzhou, Guangdong, China)

Extensive objective energy-based parameters have been measured in three auditoriums (2 scale models and 1 site survey) in the present study. Mono-aural measurements in the frequency range from 125 Hz to 4000 Hz (octave band) were conducted in unoccupied auditoriums according to the standard ISO 3382-1: 2009. Acoustical parameters, namely, the clarity C_{80}, the definition D_{50}, the early sound level Le, the late sound level Ll, the sound strength G and the center time Ts, have been estimated through analyzing the impulse responses measured in each auditorium. These parameters were spectrally averaged according to the most accepted criteria for sound quality evaluation in auditoriums and were expressed as a function of source-receiver distance. The experimental results were compared with predictions by classical and existing theoretical models proposed for concert halls and churches. A semi-empirical model based on the measured values of early and late sound levels is proposed in this work. The good agreement between predicted values and experimental data of clarity, definition, sound strength, and center time in the auditoriums analyzed indicates that the newly proposed model can be used for design predictions with reasonable accuracy.

9:15

4aAA4. Elucidation of a mechanism of acoustic impedance technique using two cardioid microphones and ensemble averaging method. Kazuma Hoshi and Toshiki Hanyu (Dept. of Architecture and Living Design, Nihon Univ., 7-24-1, Narashinodai, Funabashi, Chiba 2748501, Japan, hoshi@arch.jcn.nihon-u.ac.jp)

Recently, there is growing need for in-situ impedance measurement of several materials. We propose using two cardioid microphones (called C-C sensor) for it. C-C sensor has many advantages against two omnidirectional microphones (called P-P sensor) and one omnidirectional microphone and sound velocity (called P-U sensor). I reported the possibility of using C-C sensor for measuring acoustic impedance on 12th Western Pacific Acoustics Conference 2015 (WESPAC2015) and International Congress on Acoustics 2016 (ICA2016). When measuring some porous materials, C-C method can measure acoustic impedance as same as using P-P method, and C-C method can do it from 100 Hz to over 10 kHz. Next, we tried to elucidate the mechanism of using C-C sensor and Ensemble Averaging (EA) method using numerical experiments. As a result, the range in application of C-C sensor for measuring acoustic impedance was clarified. In this presentation, we'll introduce the measuring technique and mechanism.

9:30

4aAA5. Absorption coefficient: Dead or alive? Results of the "ASTM C423 Inter Laboratory Study". Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

In 2009 and 2011, papers were presented questioning the procedures used to measure absorption and the conversion to an "Absorption Coefficient." Part of the problem was the results of these conversions resulted in coefficients above 1.0. These results indicated a problem with the process or with the conversion to an "Absorption Coefficient." Since that time enough data have been presented to the ASTM to result in an Inter Laboratory Study (ILS) on the efficacy of ASTM C-423 and other similar worldwide standards. Many corporate and independent laboratories took part in this study and the results of this ILS will be presented in this paper. These results

helped show that the process of measuring absorption and the calculation of the "Absorption Coefficient" has had a basic flaw in it ever since it was first proposed in the 1920s. Proposed changes are discussed as well as proposed new calculations for use in calculating absorption in rooms. This should result in more accurate and consistent calculations of reverberation times for acoustic consultants. It is hoped that these results will result in a new standard for these measurements.

9:45

4aAA6. Sound absorption of Helmholtz resonators included a neck extension built in surface panel. Shinsuke Nakanishi (Dept. of Eng., Hiroshima Int. Univ., 5-1-1, Hiro-koshingai, Kure 737-0112, Japan, s-nakani@it.hirokoku-u.ac.jp)

Acoustic resonant absorber like a perforated panel or a Helmholtz resonator can be tuned at a low frequency by extending its neck or enlarging cavity volume. However, a total size of the resonators is often quite large when the neck or the cavity is simply extended for tuning at a low frequency. Previous researchers have studied Helmholtz resonator shortened in its size by subsided neck into back air cavity, and confirm that this resonator is tuned at low frequency without a deep cavity. The author has studied the effects of a winding built-in neck extension to sound absorption of perforated panels, which shows same effects as the subsided neck into back cavity. This study obtains sound absorption coefficient by measuring surface acoustic impedance at Helmholtz resonator, and discusses sound absorption of the resonator included various neck extensions built in a surface panel. Discussions in this paper focus effects of path length, patterns, or number of turns of the winding neck extension and cavity volume to the sound absorption of the Helmholtz resonator.

10:00–10:15 Break

10:15

4aAA7. A case for modular partitions in healthcare facilities. Basel Jurdy and Kathleen Gray (Acoust., Stantec, 4100 194th St. SW Ste. 400, Lynnwood, WA 98036, kathleen.gray@stantec.com)

In healthcare, good acoustic separation between rooms is critical for speech privacy and user comfort. While there are multiple factors that contribute to acoustic separation, this is usually achieved by increasing the acoustical performance of the separating architecture between rooms. This is done by running a wall of a specific Sound Transmission Class (STC) to structure to minimize flanking paths. However, running a modular wall up to or slightly above the ceiling tile can reduce the cost of construction and improve programmatic flexibility. In situations where the wall runs only to the ceiling, the acoustic ceiling tile (ACT) and plenum condition begins to play a large role in the acoustic performance of the separating architecture. Given a high-performing ACT, a high-STC modular wall, and partially open plenum, is there a condition where good speech privacy can be achieved between rooms without running walls to structure? This was examined with a mock-up for EvergreenHealth in Kirkland, WA. The results show that given certain plenum conditions and a CAC-40 tile, the separating architecture using a modular partition can perform at the minimum STC-45 separation between patient rooms set by the 2014 Facility Guidelines Institute (FGI) for healthcare facilities.

10:30

4aAA8. Objective acoustical quality in healthcare office facilities. Murray Hodgson (UBC, 2206 East Mall, Vancouver, BC V6T1Z3, Canada, murray.hodgson@ubc.ca)

Health-care facilities include many non-clinical office spaces for administrative staff; the role of acoustics in these spaces has been underexplored. This paper discusses the acoustical part of a study of indoor environmental quality (IEQ) in 17 healthcare office facilities. Physical acoustical measurements were made in six types of rooms, some with sound-masking systems, to determine the acoustical characteristics, assess their quality, relate them to the building designs, and develop prediction models. Background-noise levels were measured in the occupied buildings. In the unoccupied buildings, measurements were made of reverberation times, and "speech" levels needed to calculate speech intelligibility indices for speech intelligibility and speech privacy. In open offices, sound-level reductions per distance doubling (DL2) were measured. Noise isolations of internal partitions of different designs (double-plasterboard construction, modular or built in-situ, rising to the suspended ceiling or to the floor-ceiling slab, without and with doors, different amounts of glass) were measured. The acoustical characteristics were compared to design criteria to evaluate their acceptability. The results are presented, and are related to room type and partition design. An empirical model for predicting partition noise isolation, developed using regression techniques, is discussed. The knowledge gained from this study informs the decision-making of designers and facilities management for upgrades and future design projects.

10:45

4aAA9. Subjective acoustical quality in healthcare office facilities. Murray Hodgson (UBC, 2206 East Mall, Vancouver, BC V6T1Z3, Canada, murray.hodgson@ubc.ca)

Health-care facilities include many non-clinical office spaces for administrative staff; the role of acoustics in these spaces has been underexplored. This paper discusses the acoustical part of a study of indoor environmental quality (IEQ) in 17 healthcare office facilities. A subjective survey assessed office worker perceptions of their environments in general, and satisfaction with the acoustics. Self-reported productivity, well-being, and health outcomes were also captured. Satisfaction was lower with acoustics than with other aspects of IEQ. Satisfaction results were related to room type (e.g., open plan vs. shared vs. private office) and the absence or presence of a sound-masking system. Acoustics was the most important aspect of IEQ in predicting occupant satisfaction and well-being. Regression models were developed to predict workplace satisfaction, well-being, and job satisfaction from survey responses. Results of physical acoustical measurements showed very low correlations with subjective responses. The knowledge gained from this study informs the decision-making of designers and facilities management for upgrades and future design projects.

11:00

4aAA10. Airport Cooperative Research Project 02-51: Evaluating methods for determining interior aircraft noise levels. Randy Waldeck (CSDA Design Group, 475 Sansome St., Ste. 800, San Francisco, CA 94111, rwaldeck@csdadesigngroup.com), Paul Schomer (Schomer & Assoc., Champaign, IL), and John Freytag (Freytag & Assoc., LLC, Newport Beach, CA)

This paper documents the results of a study of acoustical testing methods used to quantify facade noise reduction measurements for airport sound insulation programs. The research was performed for the National Academy of Sciences as Airport Cooperative Research Project (ACRP) 02-51. Field measurements, in combination with acoustical calculations, were used to determine which methods are best suited for airport sound insulation programs. Loudspeaker (i.e., exterior elevated, exterior ground level, and interior), flyover, and sound intensity measurements were conducted throughout the United States. In addition, air infiltration measurements and noise reduction calculations were made. Results from each method were analyzed and compared. An analysis of each method's margin of error is presented, along with Best Practices for each method. Finally, a decision matrix for selecting the appropriate acoustical testing method is included.

11:15

4aAA11. Low frequency analysis of acoustical parameters of emotional speech for use with functional magnetic resonance imaging. Peter M. Moriarty (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, moriarty@psu.edu), Michelle C. Vigeant (Graduate Program in Acoust., Penn State Univ., State College, PA), Pan Liu, Rick Gilmore, Rachel Wolf, and Pamela M. Cole (Psych., The Penn State Univ., State College, PA)

A mother's voice plays a pivotal role in the social development of her child. Extreme exposure to vocalized anger in the home environment can weaken a child's ability to manage their emotions and potentially compromise basic societal opportunities. It is therefore critical to understand the vehicle of the emotional content in speech, known as prosody, and specifically

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the neurological interactions of a mother's voice with her child. Functional magnetic resonance imaging (fMRI) was used to measure children's brain activity while stimulated by recorded sounds of mothers speaking with an angry, happy, sad, and neutral affect. Activity was measured in response to prosodic (e.g., pitch period variations, speech rate) and voice quality features (e.g., spectral distribution, jitter). Block-style fMRI analysis often uses mean values of these acoustic features taken over the length of each utterance, usually lasting many seconds or repetition time (TR) multiples. In an effort to retain some of the time-varying prosodic information of the mothers's speech, this study included utterance-level spectral analysis of these acoustic features below the Nyquist frequency of the MRI. Preliminary results support the utility of such low-frequency analysis as a method for generating affect-differentiable information. [Work supported by NIH R21 104547.]

THURSDAY MORNING, 1 DECEMBER 2016 CORAL 2, 8:30 A.M. TO 11:35 A.M.

Session 4aAB

Animal Bioacoustics, Signal Processing in Acoustics, and Speech Communication: Sequence Information in Mammalian Vocal Call Production I

Marie A. Roch, Cochair
Dept. of Computer Science, San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720

Tadamichi Morisaka, Cochair
School of Marine Science and Technology, Tokai University, 3-20-1, Orido, Shimizu-ku, Shizuoka 4248610, Japan

Chair's Introduction—8:30

Invited Papers

8:35

4aAB1. Examining the temporal structure and information entropy of leopard seal calling bouts. John R. Buck (ECE Dept., UMass Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, johnbuck@ieee.org) and Tracey L. Rogers (School of BEES, Univ. of New South Wales, Sydney, NSW, Australia)

Leopard seals (Hydrurga leptonyx) produce sequences of stereotyped sounds, or bouts, during their breeding season. The seals share common sounds but combine them in individually distinctive sequences. This study examines the underlying structure of the calling bouts by estimating the information entropy of the sound sequences with three entropy estimators. The independent identically distributed (IID) model estimates entropy from the simple frequencies of each sound. The Markov model estimates entropy from the frequency of pairs of sounds. Finally, the nonparametric sliding window match length (SWML) estimator exploits a relationship between the information entropy and the average subsequence match length. A better model for a given sequence achieves a lower entropy estimate. This study analyzed the calling bouts of 35 leopard seals recorded during the 1992-1994 and 1997-1998 Antarctic field seasons. The decrease of entropy estimates between the IID and Markov models for all seals analyzed confirmed the presence of temporal structures in the bouts. For twenty-one of thirty-five seals, the SWML entropy estimate was not significantly less than the Markov estimate, suggesting that a first-order Markov model accurately represents the structure of their bouts. [Work supported by the Australian-American Fulbright Commission and the Australian Research Council.]

8:55

4aAB2. Sequential calls and associated behavior in captive bearded seals (*Erignathus barbatus*). Daisuke Mizuguchi (Hokkaido National Fisheries Res. Inst., National Res. and Development Agency, Japan Fisheries Res. and Education Agency, 116 Katsurakoi, Kushiro, Hokkaido 085-0802, Japan, mizgangan@gmail.com), Masatoshi Tsunokawa, Mamoru Kawamoto (Otaru Aquarium, Otaru, Hokkaido, Japan), and Shiro Kohshima (Wildlife Res. Ctr., Kyoto Univ., Kyoto, Japan)

Bearded seals produce complex underwater sounds suggested to function as territorial and/or courtship signals. However, little is known about behavioral context of the vocalization, mainly because direct observation is difficult in the wild. In this study, we recorded underwater sounds and behaviors of three captive bearded seals (an adult male and two adult females) in Otaru aquarium, Japan, between March 2012 and April 2016, in order to identify the caller and to record association between vocalization and behaviors. The male continuously vocalized from December until April with a peak in March, equivalent to breeding season in the wild. In contrast, the females vocalized only 1-2 weeks in March or April. The male and the females produced 3 and 5 call types, respectively, sharing 2 call types. The 3 call types of the male occurred in a regularly ordered sequence that was stable during this study. The females responded to the male sequential calls by nuzzling against the throat of vocalizing male during the breeding season, when the females also actively vocalized. Our results suggest that underwater vocalization of bearded seals might be used as a courtship signal between male and female possibly to advertise their reproductive status.

9:15

4aAB3. Temporal patterning differences in contact calls among odontocetes' species. Tadamichi Morisaka (School of Marine Sci. and Technol., Tokai Univ., 3-20-1, Orido, Shimizu-ku, Shizuoka 4248610, Japan, chaka@tokai-u.jp)

Contact calls in odontocetes (toothed whales) have information of individual identity or group identity. Sound types used for contact calls, however, differ among species. Signature whistles in bottlenose dolphins, which are thought to function as "names" in human, are tonal sounds, whereas PS1 calls in beluga whales are pulsed sounds. Both signature whistles and PS1 calls have usually exchange pattern that calls are followed by calls by other individuals within around 1 second. Dolphins tend to produce next call when no individuals call back. Contact calls are usually produced repeatedly in a bout. Moreover, signature whistles often include multiple loops, sometimes with breaks between loops. The duration of breaks between loops is different between bottlenose dolphins and Indo-Pacific bottlenose dolphins. These results suggest that temporal patterning in contact calls in odontocetes is important for species identity.

Contributed Papers

9:35

4aAB4. Intra-click time-frequency patterns across the echolocation beam of a beluga whale. Josefin Starkhammar (Biomedical Eng., Lund Univ., Faculty of Eng., Hoby 371, LUND 225 91, Sweden, josefin.starkhammar@bme.lth.se), Isabella Reinhold (Mathematical Statistics, Ctr. for Mathematical Sci., Lund Univ., Faculty of Eng., Lund, Sweden), Patrick Moore, Dorian Houser (National Marine Mammal Foundation, San Diego, CA), and Maria Sandsten (Mathematical Statistics, Ctr. for Mathematical Sci., Lund Univ., Faculty of Eng., Lund, Sweden)

The echolocation beam of toothed whales has been studied ever since it was first discovered in 1960. Recent studies have focused on the frequency distributions across the cross sections of the beams. Other studies have focussed on describing the entire acoustic field around the animal. However, no one has yet described the timing of each frequency component in the main lobe beam in relation to the other frequency components. Even though the echolocation clicks of broadband click species like the beluga whales (*Delphinapterus leucas*) are short in time (around 70 μs), previous results have shown indications on a frequency dependence with time, within each click. Little is known about the details of how the signal is generated and transmitted into the water. Investigations of when in time the frequency components occur within each click would give us further knowledge to how the signals are generated. This study takes a closer look at these intra-click time-frequency patterns across the echolocation beam of a beluga whale. This is done by analyzing echolocation clicks with a novel reassigned spectrogram method, developed especially for separating and determining time and frequency locations of multi-component transient signals.

9:50

4aAB5. Nonlinearities in the vocalizations of *Stenella* species in the Southwest Atlantic Ocean. Juliana R. Moron, Franciele R. de Castro, and Artur Andriolo (Laboratório de Ecologia Comportamental e Bioacústica - LABEC, Universidade Federal de Juiz de Fora, Rua José Lourenço Kelmer, Juiz de Fora, Minas Gerais, Brazil, julianamoron@hotmail.com)

Some complex features found in tonal emissions may be the result of nonlinear phenomena in sound production systems, which may result in individually acoustic signals highly variable and complex. From the analysis of five opportunistic acoustic records of distinct groups, two of spinner (*Stenella longirostris*) and three of Atlantic spotted dolphins (*Stenella frontalis*), we registered the emission of six different types of nonlinearities. Spinner dolphins had less events presenting mostly biphonation in the form of nonparallel bands and subharmonics. Atlantic spotted dolphins presented a higher rate of these features presenting mostly biphonation and screams on their vocalizations. Although the exact function of these processes is still being investigated, recent documentation in several species suggests that they may play a communicative role. Describing the occurrence of these characteristics can indicate their level of importance, as well as its structure may be indicative of its function. This is the first record of these events in *Stenella* species in the Southwest Atlantic Ocean. Future studies that allow to associate the behavior of these species while emitting these signals would greatly increase the understanding of these events. This study was funded by Chevron.

10:05–10:20 Break

Invited Papers

10:20

4aAB6. Music in the brain: The neuronal control of bird song. Michale Fee (BCS, MIT, 77 Massachusetts Ave., 46-5133, Cambridge, MA 02139, fee@mit.edu)

Whether we are speaking, swimming, or playing the piano, we are crucially dependent on our brain's capacity to reliably step through a learned sequence of states. Songbirds provide a marvelous animal model in which to study this phenomenon. Using newly developed technologies for monitoring and manipulating the activity of neurons in the brain of singing bird, we have identified circuits in the songbird brain that perform key functions of vocal production and learning. One of these circuits produces a highly precise representation of time in the brain—essentially a clock. Another circuit produces highly variable output that drives large fluctuations in the songs of juvenile birds, allowing them to creatively explore different vocal patterns during learning. I will combine these observations to propose a simple theory of how brains learn and perform such sophisticated behaviors as speech and music.

10:40

4aAB7. Neural-inspired segmentation of audio streams into phone-like units. Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720, marie.roch@sdsu.edu)

Recent work has shown that the human preauditory and/or auditory cortex is likely to play a role in acoustic landmark processing, such as the recognition of syllable and phoneme boundaries. Neurons appear to track acoustic envelopes with neural activity corresponding well with acoustic landmarks. These structures have also been observed in non-human primates, suggesting that acoustic landmark processing could be present in non-human primates and have an evolutionary role. Should landmark processing occur in non-humans, it could provide new methods for approaching animal communication. In this work, we use a simplified computer model to examine envelopes of acoustic signals that were filtered through a series of octave spaced band-pass filters. Peaks are detected in the signal envelopes with closely spaced peaks suppressed for a period of time after the first detection. Preliminary work shows an ability to detect human phoneme boundaries with a recall of 45% and a precision of 72% on the TIMIT speech corpus. The TIMIT corpus contains phoneme-level transcriptions and successful matches were defined as detecting landmarks within 20 ms of the over quarter million TIMIT phoneme boundaries. We also show qualitative results of the system for segmenting non-human primate calls into phone-like units.

11:00

4aAB8. Call concatenation in non-primate mammals. Marta B. Manser (Animal Behaviour, Univ. of Zurich, Winterthurerstr. 190, Zurich, Zurich 8057, Switzerland, marta.manser@ieu.uzh.ch)

Animals produce a variety of call sequences, from simple repetitions of the same acoustic unit to the combination of different, meaningful acoustic units resulting in a new meaning. On the example of several highly vocal mongoose species, including meerkats (Suricata suricatta), and banded mongoose (*Mungos mungo*) I identify the composition of different acoustic combinations and relate them to their function and meaning. I discuss this in regards to the underlying cognitive mechanisms and the differences and similarities in relation to other combinatorial communication systems in birds, but particularly shown for primates, and also to human language.

Contributed Paper

11:20

4aAB9. Stereotypy and variability differ between humpback whale (*Megaptera novaeangliae*) phrase types offering structural support for the hypothesis that song is multi-message display. Anita Murray, Rebecca A. Dunlop (Cetacean Ecology and Acoust. Lab., The Univ. of Queensland, Gatton, QLD 4343, Australia, anita.murray1@uqconnect.edu.au), Anne W. Goldizen (Behavioural Ecology Res. Group, The Univ. of Queensland, St Lucia, QLD, Australia), and Michael J. Noad (Cetacean Ecology and Acoust. Lab., The Univ. of Queensland, Gatton, QLD, Australia)

Humpback whale studies support inter- and/or intrasexual song functions, suggesting dual functionality. Some animals produce multi-message displays with dual functionality. Stereotyped components, for mate attraction and/or male-male interactions, convey species/group membership. Variable components, for courtship, convey male quality. If humpback whale song is a multi-message display, then song should include both stereotyped and variable components. Recordings of east Australian males (2004:8, 2011:9) were used to investigate this hypothesis. Song comprises "units" arranged into sequences ("phrases") that are repeated ("themes"). Themes are characterized by different phrase types. Variation among phrases repeated within a theme ("phrase variants") is mostly overlooked by other studies. Whether some phrase types are more variable than stereotyped is unknown. To assess this, repertoire size (number of unit types and phrase variants) was determined per phrase type. Whether phrase variants were unique to individuals or shared were also determined. Unit types were defined by self-organizing maps. Levenshtein distances between phrases with cluster analyses determined phrase types and phrase variants illustrating a novel application of these methods. Stereotypic phrase types contained shared phrase variants and limited repertoire. Variable phrase types contained unique phrase variants and large repertoire. These results support the hypothesis that song is a multi-message display.

Session 4aAO

Acoustical Oceanography, Animal Bioacoustics, and Signal Processing in Acoustics: Acoustic Scattering by Aquatic Organisms I

Kelly J. Benoit-Bird, Cochair
College of Earth, Ocean, and Atmospheric Sciences, Oregon State University, 104 COEAS Admin Bldg., Corvallis, OR 97331

Kouichi Sawada, Cochair
Fisheries Technology, National Research Institute of Fisheries Engineering, FRA, 7620-7, Hasaki, Kamisu 3140408, Japan

Timothy K. Stanton, Cochair
Woods Hole Oceanographic Institution, Dept. Appl. Ocean. Phys. & Eng., Woods Hole Oceanographic Institution, Woods Hole, MA 02543

Chair's Introduction—8:15

Invited Papers

8:20

4aAO1. Emergent pattern and underlying process: The what, why, and how of fish schooling. Julia K. Parrish (School of Aquatic and Fisheries Sci., Univ. of Washington, 1492 NE Boat St., Seattle, WA 98195, jparrish@uw.edu)

Over half of the world's fish aggregate at some point in their lives. As larvae, juveniles and/or adults, fish assemble into dense three-dimensional groups that last hours to lifetimes. The largest single stock fisheries in the world—anchoveta in Peru and pollock in the Bering Sea—are schooling fish. At several times the entire world's fish catch, the myctophid aggregations making up the deep scattering layer form the world's largest school. But are all schools alike? Are the "rules of fish schooling" at the scale of the individual the same across all species? Do different rules produce similar emergent pattern scaled up to the group, population, and ecosystem? Could differences across these scales from individual behavior to population response be determined acoustically, allowing technology to inform behavior, ecology, and conservation? This talk explores the diversity of ways that fish assemble, fiss and fuse, and disaggregate on a daily, seasonal, and annual basis through three related lens: the patterns of schooling, reasons why fish school, and analysis of how fish maintain the physical structure of the school.

8:40

4aAO2. Coastal observation systems to monitor physical, chemical, and biological parameters. Hidekatsu Yamazaki (Dept. of Ocean Sci., Tokyo Univ. of Marine Sci. and Technol., 4-5-7 Konan, Minato, Tokyo 108-8477, Japan, hide@kaiyodai.ac.jp), Eiji Masunaga (Ibaraki Univ., Ibaraki, Japan), Scott Gallager (Woods Hole Oceanogr. Institution, Woods Hole, MA), Mamoru Tanaka, Marika Takeuchi, Kazuo Amakasu (Dept. of Ocean Sci., Tokyo Univ. of Marine Sci. and Technol., Tokyo, Japan), and Hayato Kondo (Dept. of Maritime System Eng., Tokyo Univ. of Marine Sci. and Technol., Tokyo, Japan)

We have developed a free-fall multi-parameter profiler (YODA Profiler) to measure various physical and biological parameters in coastal ocean. We found internal bores create a strong mixing event. Sediment resuspension is associated with the mixing event and also AZFP detected fish school at the front of bore. We have deployed a cable observatory system (Oshima Coastal Environment data Acquisition Network System, OCEANS) in a coastal area to monitor coastal ecosystem continuously. OCEANS can measure various physical, chemical and biological parameters simultaneously, and operates a plankton imaging system (Continuous Plankton Imaging and Classification System, CPICS). Based on acquired images of phytoplankton and zooplankton, we are investigating how planktonic biodiversity is affected by multi-scale physical processes, such as Kuroshio and internal waves. We are developing a technique to predict the biodiversity of plankton from three-dimensional hydrodynamic model using a newly developed plankton ecosystem model. We are also developing a AUV (MEMO-pen) that carries CPICS as well as microstructure profiler (TurboMAP) in order to simultaneously observe turbulence and plankton. We will introduce the most recent results from these observation systems.

9:00

4aAO3. Seasonal and interannual changes in the sound scattering layer at deep-sea floor in the Amundsen Sea, Antarctica. Hyoung Sul La, Keyhong Park (Korea Polar Res. Inst., Songdo-miraero 26, Yeonsu-gu, Incheon 406-840, South Korea, hsla@kopri.re.kr), Anna Wåhlin (Univ. of Gothenburg, Gothenburg, Sweden), Ho Kyung Ha (Inha Univ., Incheon, South Korea), Angus Atkinson (Plymouth Marine Lab., Plymouth, United Kingdom), Sophie Fielding (Br. Antarctic Survey, Cambridge, United Kingdom), Dong Sun Kim (Korea Inst. of Ocean Sci. and Technol., Ansan, South Korea), Eun Jin Yang, Tae Wan Kim, SangHoon Lee (Korea Polar Res. Inst., Incheon, South Korea), JungHo Im (Ulsan National Inst. of Sci. and Technol., Ulsan, South Korea), and Hyoung Chul Shin (Korea Polar Res. Inst., Incheon, South Korea)

Vertical migration of zooplankton is ubiquitous behavior in marine plankton community; however, seasonal and interannual behavior are little observed in the deep sea under seasonal varying sea ice. Here, the first evidence that sound scattering layers of zooplankton can support the knowledge for understanding the effect of climate change is presented, based on four-years acoustic backscattering strengths in the Amundsen Sea, Antarctica. Amundsen Sea is a biological hotspot region with the rapid oceanic melting of the ice shelf as well as the most productive (per unit area) in the Antarctic. High-temporal resolution profiles of acoustic backscattering strength collected from a bottom-moored, upward looking Acoustic Doppler Current Profiler were examined to describe the temporal variation of sound scattering layers. Our observations show that sound scattering layers exhibited clear diel, seasonal, and interannual pattern associated with solar radiation, sea ice concentration, and phytoplankton biomass. The timing of seasonal variation is also closely related with climate index for Southern Annular Mode and El Niño Southern Oscillation. This observation points out that acoustic signals from sound scattering layers could be a proxy to understand zooplankton ecosystem response to climate shifts.

9:15

4aAO4. Development and field application of early warning system for harmful algal blooms (red-tide) early warning using ultrasound wave, Korea. Donhyug Kang, Hansoo Kim, Seungwon Jung, Mira Kim (Korea Inst. of Ocean Sci. & Technol., 787 Haeanro, Ansan, Seoul 15627, South Korea, dhkang@kiost.ac.kr), and Byoung-Kweon Kim (Syscore Inc., Seoul, South Korea)

The harmful algal blooms (HABs) are affecting the ecosystems, coastal fisheries, and social economics in Korea. In the case of 2013 summer season, massive damage due to HABs (species: *Cochlodinium polykrikoides*) was reached to $23 million during August, 2013 in Korea. The most important things for minimizing of HABs damage is rapidly detection in initial stage. For this purpose, we have developed a real-time HABs detection system using ultrasound. The integrated system is composed by several sensors (3.5 MHz frequency, temperature and salinity, Chlorophyll, Dissolved oxygen), power supply, network communication, and remote control modules. For evaluating performance of the developed system, several experiments were intensively implemented in the laboratory and *in-situ* environments during summer seasons, 2013-2016. As a result, both of the performance tests are confirming the acoustic backscattering signal increases with the number of *C. polykrikoides*. Furthermore, the integrated system shows successfully *in-situ* performance for real-time red-tide detection and corresponding environmental data.

9:30

4aAO5. Comparison of the volume backscattering strength measured by EK60 and EK80. Tohru Mukai (Faculty of Fisheries Sci., Hokkaido Univ., 3-1-1 Minato, Hakodate, Hokkaido 0418611, Japan, mukai@fish.hokudai.ac.jp) and Kazuo Amakasu (Tokyo Univ. of Marine Sci. and Technol., Tokyo, Japan)

Walleye pollack (Gadus chalcogrammus) resources are abundant around Hokkaido in Japan. The acoustic survey of the walleye pollack using the quantitative echo-sounder has been performed around Hokkaido since 1995. The quantitative echo-sounder that is mainly used for this survey is the EK60 system consisting of GPT + ER60 software made by Simrad. On the other hand, EK80 system using the WBT has been released in 2015, and the acoustic survey will be conducted by EK80 system from now. Therefore, it is necessary to examine whether the acoustic data measured by the EK60 and EK80 systems are identical. In this study, in the area where sound scattering layer was seen, these two systems were used alternately on the same track line to examine whether there are differences in the acoustic data to be measured. Systems used in this study were GPT in frequency 38, 120, and 200 kHz, controlled by ER60 (Ver.2.4.3) and WBT in CW mode at 120, 200 kHz controlled by EK80 (Ver.1.8.3). The acoustic data of the same transect were collected by both systems. The difference of SV between both systems will be discussed.

9:45

4aAO6. Size estimation of walleye pollack (*Theragra chalcogramma*) by using a broadband split-beam system. Tomohito Imaizumi (National Research Inst. of Fisheries Eng., Japan Fisheries Res. and Education Agency, 7620-7 Hasaki, Kamisu-shi 314-0408, Japan, imat@affrc.go.jp), Kazuhiro Sadayasu (Marine Fisheries Res. and Development Ctr., Japan Fisheries Res. and Education Agency, Yokohama-shi, Japan), and Tomonari Akamatsu (National Res. Inst. of Fisheries Sci., Japan Fisheries Res. and Education Agency, Yokohama-shi, Japan)

Recently, broadband quantitative echo sounders have been developed. These systems can measure single echo traces from individual fish with high range resolution. They can be used for the accurate size distribution by measuring the target strength. On the other hand, walleye pollack is one of the important species for stuck management in Japan. They form a school at juvenile stage. We compared the captured fish size using bottom trawl net with the acoustically estimated values. We conducted the survey in July 2015 in east side of off Hokkaido, Japan, using the fisheries research vessel Kaiyo-maru No. 5 (495 tons), which is equipped with narrowband echo sounder transducers (EK-60, Kongsberg, 38, 70, 120, and 200 kHz). The broadband split-beam transducer (FSS-SBX, Furuno) was deployed at the ship sideboard. All of the clock of the system were synchronized for comparison of data. More single echo traces could be measured using the broadband system than those measured by the narrowband system. Fork lengths were estimated at 3~70 cm based on the maximum normalized target strength value: -60 dB. We confirmed that good agreement of size distribution between catch and acoustic estimation. The broadband systems may be helpful tool for estimating body size of fish.

10:00

4aAO7. Potential for broadband acoustics to improve stock assessment surveys of midwater fishes. Christopher Bassett (NOAA Fisheries, Alaska Fisheries Sci. Ctr., National Marine Fisheries Service, Alaska Fisheries Sci. Ctr., Resource Assessment and Conservation Eng., 7600 Sand Point Way NE, Seattle, WA 98115, chris.bassett@noaa.gov), Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH), Chris Wilson, and Alex De Robertis (NOAA Fisheries, Alaska Fisheries Sci. Ctr., Seattle, WA)

Acoustic-trawl surveys rely on a combination of backscatter measured with echosounders and species composition data from trawls to apportion the backscatter to different species and size classes. Narrowband echosounders have been widely used in this context for decades. Multi-frequency analysis of narrowband echosounder data has been shown to be effective for discriminating between diverse taxa (e.g., euphausiids vs. swimbladdered fishes) but distinguishing morphologically similar species (e.g., swimbladdered fishes) remains a major challenge. Previous work indicates that broadband backscatter techniques have the potential to improve such acoustic target characterizations by exploiting nearly continuous frequency spectra, but these methods have not been widely applied in fisheries surveys. The recent commercial availability of broadband transceivers is accelerating the evaluation of this technology. We present operational data from two broadband acoustic scattering systems: (1) 14-160 kHz and (2) 3-10 kHz used during surveys of walleye pollock (*Gadus chalcogrammus*) in the Gulf of

Alaska and Eastern Bering Sea. The presentation focuses on the potential for: 1) discrimination among common species in the area and 2) utilization of swimbladder resonance to estimate the dominant size class in single-species fish aggregations. The implications for fish stock assessment surveys are considered.

10:15–10:30 Break

10:30

4aAO8. Broadband acoustics: A viable tool for quantification of cod egg density and distribution? Gavin Macaulay, Egil Ona, Rokas Kubilius, and Olav Rune Godø (Marine Ecosystem Acoust., Inst. of Marine Res., C Sundts Gate 64, Bergen 5817, Norway, gavin.macaulay@imr.no)

Fish eggs are ichthyoplankton with a similar behavior in the water column as small zooplankton. Their specific density relative to sea water determines their vertical distribution and spread; horizontal transport of the layers will then determine their fate and geographical position during the hatching of the larvae. Acoustic techniques are routinely used to assess the quantities and distribution of fish populations. In comparison to fish, zooplankton and ichthyoplankton are very weak acoustic reflectors and this makes the application of acoustic techniques challenging. We assessed the potential feasibility of using broadband acoustics to detect cod eggs in a two stage process: first, we carried out small-scale tank experiments to measure the acoustic reflectivity of individual and grouped cod eggs over the 170 to 360 kHz frequency range. These experiments showed that natural densities of cod eggs could be detected above the background acoustic noise level. This lead to the second stage whereby we conducted a field test in an area with known concentrations of cod eggs, using a combined acoustic-optical towed and moored system. We postulate that the high spatial resolution, yet rapid wide area coverage possible with acoustics will lead to a better description of egg distributions and subsequent larval distributions and help to understand the extremely variable recruitment success in sub-arctic and arctic ecosystems.

10:45

4aAO9. Mid-frequency clutter and reverberation characteristics of fish in a shallow ocean waveguide. Wu-Jung Lee, Dajun Tang, and Eric I. Thorsos (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu)

Horizontal-looking, mid-frequency sonar systems allow synoptic underwater observation over kilometer scales and are useful for a wide range of applications. However, quantitative assessment of fish aggregations using such systems in a shallow water environment is challenging due to the complexity of sound interactions with the surface and seafloor. Based on data collected during TREX13 (Target and Reverberation Experiment 2013), this study investigates methods to reliably distinguish fish echoes from background contributions to reverberation. The data were collected on a fixed, horizontal receiving line array from a source transmitting linear frequency-modulated signals in the band between 1.8 and 3.6 kHz. The experimental site was nearly isodepth (approximately 20 m) and characterized by a well-mixed, isothermal water column. Fish echoes were ubiquitously found in the data, with noticeable differences between day and night. In a particularly interesting case, a large aggregation of fish was observed emerging from a shipwreck, evolving in space, obscuring the wreck echoes temporarily, and eventually dispersing and disappearing into the overall reverberation background. Using a physics-based approach, the scattering level, statistical features, and spatial characteristics of both the fish echoes and background reverberation in this data set were analyzed and modeled. [Work supported by the APL postdoctoral fellowship and ONR.]

11:00

4aAO10. Heterogeneity of deep scattering layer shapes the Bahamian mesopelagic ecosystem. Mei Sato and Kelly J. Benoit-Bird (Oregon State Univ., 104 CEOAS Admin Bldg., Corvallis, OR 97331, msato@coas.oregonstate.edu)

We explored deep scattering layers off the Bahamas where the beaked whales have been historically observed. By comparing the habitats frequently used by the beaked whales with the ones rarely utilized, we examined the differences in mesopelagic ecosystem potentially driving the whales' prey habitat through bottom-up control. Using ship-based multifrequency echosounders and direct net sampling, we identified common layer structures characterized by the diffuse, broad layers (>100 m in thickness) observed across the study site. Within those diffuse layers, we occasionally observed distinctively bounded intense layers (~20 m in thickness) located at the upper edges. Instead of a specific layer structure dominating the habitat characteristics, we found that heterogeneity of the layer structures shaped the preferred habitat of beaked whales. Frequently used habitat was characterized by evenly distributed diffuse layers alone, or combined with patchy thin layers. Less-used habitat showed diverse characteristics such as diffuse layers combined with highly variable patchiness of the thin layers. The layer depth, its thickness, and frequency response were not related to the preference of the habitats. Heterogeneity of the layer structures reveals the dynamics of the mesopelagic ecosystem, potentially impacting higher trophic levels.

11:15

4aAO11. Comparison of zooplankton density estimated by acoustic inversion method and net sampling. Kouichi Sawada (Fisheries Technol., National Res. Inst. of Fisheries Eng., FRA, 7620-7, Hasaki, Kamisu, Ibaraki 3140408, Japan, ksawada@fra.affrc.go.jp), Tohru Mukai (Faculty of Fisheries Sci., Hokkaido Univ., Hakodate, Hokkaido, Japan), Yoshiaki Fukuda (Bluefin Tuna Resources, National Res. Inst. of Far Seas Fisheries, FRA, Shizuoka, Shizuoka, Japan), and Tomohiko Matsuura (Fisheries Technol., National Res. Inst. of Fisheries Eng., FRA, Kamisu, Ibaraki, Japan)

A multi-frequency acoustic zooplankton fish profiler (AZFP, ASL Environmental Sciences) with temperature-salinity and depth sensor was moored at 10 m from the bottom (ca. 30 m) at the mouth of Yamada Bay in Iwate Prefecture, Japan, from 14 January to 16 June 2015 in order to measure volume backscattering strength profiles from the transducer surface to the water surface at four frequencies (125, 200, 455, and 769 kHz). Zooplankton density by size was estimated from the measured volume backscattering strength at four frequencies by the inversion method. Monitoring period was selected to cover the period when juvenile salmon stay in the bay before leaving for oceans. Zooplankton sampling was conducted periodically (once a week or two weeks, day and night) using a Kitahara Net (30 cm dia., 335 çm mesh) and a Norpac Net (45 cm dia., 335 çm mesh) at near the moored position. These observations revealed that large zooplankton, which juvenile salmon prefers, arrived with the Oyashio Current, and the zooplankton density increased from late April to the middle of May. Acoustically estimated variation pattern in zooplankton density over time was similar to that obtained by net sampling. [This work was supported by AFFRC, Japan.]

11:30

4aAO12. Effects of target strength variability on estimates of abundance: The case of Atlantic mackerel (*Scomber scombrus*). Ben Scoulding (Echoview, GPO Box 1387, Hobart, TAS 7001, Australia, ben.scoulding@echoview.com), Sven Gastauer (Ctr. for Marine Sci. and Technology, Curtin Univ., Perth, WA, Australia), David MacLennan (Inst. of Biological and Environ. Sci., Aberdeen Univ., Aberdeen, United Kingdom), Sascha Fässler (IMARES, IJmuiden, Netherlands), Phillip Copland (Marine Scotland Sci., Aberdeen, United Kingdom), and Paul Fernandes (Inst. of Biological and Environ. Sci., Aberdeen Univ., Aberdeen, United Kingdom)

Atlantic mackerel is a small pelagic, migratory fish which supports valuable commercial fisheries. Despite the fact that these fish school in massive numbers, and are readily detected using echosounders, fishery-independent estimates of the abundance of mackerel in the Northeast Atlantic do not yet consider acoustic data. Echo-integration surveys could provide annual estimates of abundance, with additional scope for studying mackerel distributions throughout the year. However, as in all acoustic surveys, this requires accurate estimates of target strength (TS). The present study provides *in situ* TS estimates for mackerel from measurements made at sea with a multi-frequency split-beam echosounder. Empirical results suggest mean TS of -51.22 dB at 18 kHz, -59.83 dB at 38 kHz, -55.51 dB at 120 kHz, and -53.43 dB at 200 kHz, for a mean fish length of 33.3 cm. These differ significantly from the values currently used in acoustic surveys. The sensitivity of mackerel abundance estimates to variations in TS estimates was investigated

using data from a dedicated mackerel acoustic survey. Total stock biomass estimates at 38 and 200 kHz were in very good agreement with each other (to within 2.2%) and were in the range of values from an independent (i.e., non-acoustic) mackerel stock assessment. It is recommended that future acoustic estimates of mackerel are based on integration at 200 kHz.

11:45

4aAO13. Using echoview software to characterise acoustic scattering by aquatic organisms: The case of Atlantic mackerel (*Scomber scombrus*). Toby Jarvis and Ben Scoulding (Echoview, GPO Box 1387, Hobart, TAS 7001, Australia, toby.jarvis@echoview.com)

In a separate presentation, Scoulding *et al*. describe the rationale and foundation for regular acoustic surveys of Atlantic mackerel (*Scomber scombrus*) in the northeast Atlantic Ocean. Two key aims of this work were (1) to obtain a good knowledge of the *in situ* target strength of these weak scatterers, and (2) to settle on a robust and efficient workflow for the processing of mackerel acoustic survey data. To achieve these aims, dedicated mackerel acoustic surveys were carried out in 2012 and 2014 around the Shetland Islands, Scotland, and the data were processed using Echoview software. In this software-focused presentation, we provide details on how the multifrequency single-beam echosounder data were processed, and we discuss this in the context of a general workflow for acoustic data processing that can be considered for any type of echosounder and application. The aim is to provide specific usage examples of some of the powerful features of Echoview and to provide general guidance for those who use or who are considering using Echoview to characterise acoustic scattering by aquatic organisms.

THURSDAY MORNING, 1 DECEMBER 2016 — CORAL 1, 7:30 A.M. TO 11:00 A.M.

Session 4aBAa

Biomedical Acoustics: Session in Honor of Floyd Dunn

W. D. O'Brien, Jr., Cochair
University of Illinois, 405 N. Mathews, Urbana, IL 61801

Yoshifumi Saijo, Cochair
Tohoku University, 4-1 Seiryomachi, Aoba-ku, Sendai 980-8575, Japan

Chair's Introduction—7:30

Invited Papers

7:35

4aBAa1. Celebration of Floyd Dunn: Some remarks about his life, career, and accomplishments. W. D. O'Brien, Jr. (Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801, wdo@uiuc.edu)

No Celebration of Floyd Dunn would be complete without some remarks about the life, career, and accomplishments of this remarkable scientist/engineer. 1946 is a key year in the history when Floyd arrived at the University of Illinois as an undergraduate student, having served in the European Theater during World War II. The same year, William J. (Bill) Fry and Francis J. (Frank) Fry were hired by the University of Illinois, and the Bioacoustics Research Laboratory was established (70 years ago!). Floyd had made significant scholarly contributions in six general themes. The six ultrasonic biophysics themes include: absorptive processes, nonlinear phenomena, application in living systems, toxicity, measurement techniques, and ultrasonic microscopy. Floyd has been recognized for his accomplishments by being the recipient of most the important national and international awards including: National Academy of Sciences, National Academy of Engineering, ASA's Gold Medal and Silver Medal in Bioresponse to Vibration, Acoustical Society of Japan's Medal of Special Merit, IEEE's Edison Medal, and election to Fellowship in six professional societies.

7:55

4aBAa2. History of biomedical ultrasound microscope in Japan. Yoshifumi Saijo (Tohoku Univ., 4-1 Seiryomachi, Aoba-ku, Sendai 980-8575, Japan, saijo@idac.tohoku.ac.jp)

Floyd Dunn greatly contributed to the history of biomedical ultrasound microscope (USM) in Japan. Ten years after the concept of ultrasound microscope was proposed by Sokolov of Soviet Union, he realized the ultrasound absorption microscope in 1959. During the time he directed the Bioacoustics Research Laboratory at the University of Illinois, many Japanese researches stayed in his laboratory. Especially, the researchers from Tohoku University learned both technology and philosophy of USM and they started development of scanning acoustic microscopy for medicine and biology in 1982. The USM system had been used for nearly 30 years until the Great East Japan Earthquake destroyed the system. Not only USM, other high frequency ultrasound imaging was also developed in 1990s. Especially, clinical application of intravascular ultrasound filled a gap between several MHz ultrasound used in conventional B-mode and 100 MHz used in USM. High frequency technology and computer science helped developing new generation USM such as ultrasound impedance microscope and 3D ultrasound microscope. Now the application area is spread and Biomedical Ultrasound Microscope Conference is biannually held in Japan. Now, we thank Floyd again for the fruitful activity of USM in Japan.

8:15

4aBAa3. Non-invasive *in-situ* quantitative observation of biological cells by acoustic microscopy. Naohiro Hozumi, Kyoichi Takanashi, Mamoru Washiya, Sachiko Yoshida (Toyohashi Univ. of Technol., 1-1 Hibarigaoka, Tenpaku, Toyohashi, Aichi 441-8580, Japan, hozumi@icceed.tut.ac.jp), Kazuto Kobayashi (Honda Electronics, Tohohashi, Japan), and Yoshifumi Saijo (Tohoku Univ., Sendai, Japan)

Several ways of quantitatively observing biological cells will be exhibited. Cells are cultured on a thin plastic film in order to make is possible to access by high-frequency acoustic beam focused onto the cell. Reflection waveform has a spectrum briefly spreading from 100 to 400 MHz. Mechanical scan makes it possible to acquire a 2-D profile. Cells in contact with the substrate can be characterized by means of characteristic acoustic impedance. In most of observations, buffer liquid that is simultaneously observed with cells may be employed as the reference material for calibration. In the process of cell culture, some of the cells may be delaminated from the substrate. In such a case significant error in acoustic impedance measurement may take place. In order to avoid such an error waveform separation is performed. As a result of such a waveform analysis, delamination distance may be calculated as well as the acoustic impedance of the cell that is slightly distant from the substrate. As an example C2C12 cells that develop into muscle tissue were observed after differentiation-inducing. Finally, feasibility of cross-sectional observation of cells will be discussed by showing some examples.

8:35

4aBAa4. Evolution of acoustic microscopy in the commercial marketplace. Lawrence W. Kessler (Sonoscan Inc., 2149 E. Pratt Blvd., Elk Grove Village, IL 60007, lkessler@sonoscan.com)

The promise of Acoustic Microscopy from the 1970s,—to image features and structures of samples from a micromechanical aspect—has found a few niche markets in industry, and, in particular, defect detection in manufactured products. The hope for its widespread application in medicine and pathology has not yet been reached in the commercial market, though there are several pockets of significant biomedical research that employ acoustic microscopy methods. This paper will give this author's perspective on the field from his 40 years of experience at Sonoscan, Inc., a company originally founded to explore commercial applications, and whose primary activities have found to be serving the semiconductor and microelectronics industries. Based on the original Ultrasonic Absorption Microscope paper by Dunn and Fry in 1959, the early commercial technique substituted a scanning laser as an acoustic detector for their thermocouple probe to produce two dimensional acoustic images.

8:55

4aBAa5. At the Intersection: Science, fatherhood, and historical events. Unscientific remarks about Floyd Dunn through the lens of his career at the meeting point of historical events, fatherhood, and possibility. Roo Dunn (None, 57 Green St., Bath, ME 04530, dunnroo@gmail.com)

Floyd's 60 years of scientific contributions to ultrasonic biophysics and his life experiences created, and crossed with, many of the thrilling and tumultuous events of the 20th and early 21st centuries. We will consider national and global events in economics, war, geopolitics, cosmology, and black holes. This paper will look at these topics through conversations spanning some 40 years. We will examine how they meshed with his life, family and the hypotheticals that were frequently discussed which might have drastically altered the outcome. The material is based on informal talks between father, son, and youngest grandson.

9:15

4aBAa6. Application of scanning acoustic and photoacoustic microscopes in orthopedic surgery. Yoshihiro Hagiwara (Dept. of Orthopaedic Surgery, Tohoku Univ. School of Medicine, 2-1 Seiryo-machi, Aoba-ku, Sendai 980-8574, Japan, hagi@med.tohoku.ac.jp) and Yoshifumi Saijo (Dept. of Biomedical Imaging, Graduate School of Biomedical Engineering, Tohoku Univ., Sendai, Japan)

Demands for orthopedic surgery, decreasing pain and disabilities, are increasing in developed countries with older nations. Together with increasing number of patients, its cost is now rising. Among modalities for noninvasive diagnosing techniques, such as plain X-rays, computed tomography, magnetic resonance imaging (MRI), and ultrasonography (US), MRI is a powerful tool for evaluating anatomical abnormalities. However, location and economic limitations for routine use remain. US is quick and inexpensive with a higher resolution than MRI. US plays an important role in assessing musculoskeletal soft tissues. A scanning acoustic microscope (SAM) characterizes biological tissues by estimating the elastic parameters based on sound speed. Biomedical photoacoustic (PA) imaging has the unique capability of combining high optical contrast and high ultrasound resolution in a single modality. Osteoarthritis and frozen shoulder are the major problems in musculoskeletal diseases. Osteoarthritis is a degenerative joint disorder characterized by the progressive degeneration of articular cartilage, osteophyte formation, and subsequent joint space narrowing. Further, frozen shoulder is characterized with severe pain and a decrease in shoulder motion, which is caused by joint capsular stiffness. We applied SAM and PA to animal models for assessing.

9:35–9:50 Break

9:50

4aBAa7. Low-energy extracorporeal shock wave therapy, preclinical and clinical studies. James F. Greenleaf, Carolina A. Carrascal, Matthew W. Urban, Randy Kinnick, Amir Lerman, and lilach Lerman (Biomedical Eng., Mayo, 200, First ST SW, Rochester, MN 55901, jfg@mayo.edu)

Floyd Dunn's extensive research career contributed significantly to safety and efficacy of diagnostic and therapeutic ultrasound. Positive effects of low-energy shock wave therapy (SWT) show preclinical promise in renal disease in pigs and in cardiovascular disease in humans. Preclinical: a set of 26 pigs were divided into four groups, normal + SWT, atherosclerotic renal artery stenosis (ARAS) + SWT,

and Normal and ARAS pigs. After three weeks of ARAS or sham, SWT was applied twice a week for three weeks. SWT after ARAS decreased blood pressure, recovered the stenotic microvascular density, improved renal blood flow and glomerular filtration rate, and decreased fibrosis. No noticeable kidney injury was observed. Low-energy SWT recovers the stenotic kidney toward normal function by preserving intra-renal microcirculation and may also alleviate renovascular hypertension. 72 patients with refractory angina, 43 were treated with low-energy SWT in the area peripheral to their infarct and 29 patients were in the control group. No adverse effects were seen in the treated group. At six months post-treatment angina class score was decreased ($p = 0.0002$), nitroglycerine use was reduced ($p = 0.03$), and hospitalization rate was reduced ($p = 0.03$). Low-energy shockwave therapy may be a tool for treating diseases caused by ischemia.

10:10

4aBAa8. Cavitation in therapeutic ultrasound: To be avoided or to be utilized. Shin-ichiro Umemura (Graduate School of Biomedical Eng., Tohoku Univ., Aoba 6-6-05, Aramaki, Aoba-ku, Sendai 980-8579, Japan, sumemura@ecei.tohoku.ac.jp), Shin Yoshizawa, and Ryo Takagi (Graduate School of Eng., Tohoku Univ., Sendai, Miyagi, Japan)

Nearly three decades ago, I spent a kind of sabbatical year in Bioacoustic Research Laboratory, University of Illinois, which was supervised by Professor Floyd Dunn. Human knowledge and technologies on acoustic cavitation were significantly limited at that time compared to now. As the result, acoustic cavitation was considered to be avoided even in therapeutic applications of ultrasound. However, I was thinking that it should be utilized and had discussions with him on it, who was the expert on bioeffects of ultrasound including those induced by cavitation. Human technologies on acoustic cavitation have been advanced since then mainly in two aspects: 1) ultrasonic as well as optical observation of cavitation microbubbles at a high frame rate, and 2) generation of a short ultrasonic pulse at extremely high amplitude. A high speed camera had made possible to observe that a short extremely high-amplitude pulse can generate cavitation microbubbles localized near the ultrasonic focal point without inducing their chaotic generations at least in water. High frame rate ultrasonic imaging have made possible to confirm the localization in biological tissue during the therapeutic focused ultrasound treatment. *In-vivo* as well as ex-vivo experimental results evidencing those will be shown and discussed in the paper.

Contributed Papers

10:30

4aBAa9. Cumulative effect in the chronotropic effect in rat hearts caused by pulsed ultrasound. Olivia C. Coiado (Donald P. Shiley School of Eng., Univ. of Portland, 5000 N Wilamette Blvd., Portland, OR 97203, coiado@up.edu) and William D. O'Brien, Jr. (Univ. of Illinois, Urbana, IL)

This study investigated the cumulative effect of a decreasing sequence of pulse repetition frequencies (PRFs) on the chronotropic effect via the application of 3.5-MHz pulsed ultrasound (US) on the rat heart. Two 3-mo-old female rat groups ($n = 4$ ea) were studied: control (US off) and PRF decrease. Rats were exposed to transthoracic ultrasonic pulses at ~0.25% duty factor at 2.0-MPa peak rarefactional pressure amplitude. Three PRF sequences were applied. The PRF started greater than that of the rat's heart rate and was decreased sequentially in 1-Hz steps every 10 s (i.e., 6, 5 and 4 Hz—one sequence) for a total duration of 30 s. The total US exposure was 90 s (3-sequences). For the PRF decrease group, the ultrasound application resulted in an ~8% significant negative chronotropic effect after the first PRF sequence, ~9% significant negative chronotropic effect after the second PRF sequence, and ~10% significant negative chronotropic effect after the third PRF sequence. No significant changes were observed for the control group. The ultrasound application caused a negative chronotropic effect after US exposure for the PRF decrease group and a slight cumulative effect was observed. [Support: NIH Grant R37EB002641.]

10:45

4aBAa10. The interaction of pulsed ultrasound with mammalian lung. Douglas Miller (Radiology, Univ Michigan, 3240A Medical Science I, 1301 Katherine St., Ann Arbor, MI 48109-5667, douglm@umich.edu)

The impedance difference for a water-air interface yields virtually complete reflection for ultrasound, which coincidentally delivers an acoustical radiation surface pressure (ARSP). ARSP may be a physical mechanism for pulmonary capillary hemorrhage induced by diagnostic ultrasound. Fundamental tissue parameters in biomedical acoustics are the density ρ, speed of sound **c**, and the resulting acoustical impedance $\mathbf{z}=\rho\mathbf{c}$. Many tissues are similar to water ($\mathbf{z}$ = 1,628 krayl), but not lung. The gas content of lung imparts an extreme heterogeneity with air having $\mathbf{z}$ = 0.424 krayl. If the lung surface is modeled simply as a water-air interface, then the ARSP is maximal. During a diagnostic ultrasound pulse, ARSP can be comparable to the pulmonary capillary blood pressure, thus adding stress to capillaries. However, the bulk properties of the lung are different from air. One of the few reports on acoustical properties of lung was by Floyd Dunn (JASA 1986;80:1248). Consideration of these bulk properties of lung tissue yields a much higher $\mathbf{z}$ $\approx$ 336 krayl at 2.8 MHz, and indicates substantial transmission of ultrasound through the pleura (O'Brien *et al.* JASA 2000;108:1290). This model suggests a reduced ARSP, and the potential involvement of additional mechanisms for capillary injury within the pulmonary interior.

Session 4aBAb

Biomedical Acoustics: Medical Acoustics in Kidney and Liver Disease I

Norihiro Koizumi, Cochair
Graduate School of Informatics and Engineering, The University of Electro-Communications (UEC), 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585 JAPAN, Chofu 182-8585, Japan

Michael Bailey, Cochair
Center for Industrial and Medical Ultrasound, Applied Physics Lab., University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Hiroyuki Fukuda, Cochair
Yokohama City Univ. Medical Center, Yokohama, Japan

Adam D. Maxwell, Cochair
University of Washington, 1013 NE 40th St., Seattle, WA 98105

Chair's Introduction—11:15

Contributed Paper

11:20

4aBAb1. Quantitative fibrotic imaging based on multi-Rayleigh model for ultrasound B-mode image of liver fibrosis. Shohei Mori, Shinnosuke Hirata (Systems and Control Eng., Tokyo Inst. of Technol., 2-12-1 S5-17, Ookayama, Meguro-ku, Tokyo 152-8552, Japan, mori@us.ctrl.titech.ac.jp), Tadashi Yamaguchi (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba, Japan), and Hiroyuki Hachiya (Systems and Control Eng., Tokyo Inst. of Technol., Tokyo, Japan)

A quantitative diagnostic method for liver fibrosis using ultrasound B-mode image is highly required. To quantitatively evaluate liver fibrosis using ultrasound B-mode image, we have been focusing on the echo amplitude distribution properties. A probability density function (PDF) of echo envelope from a normal liver can be approximated by a Rayleigh distribution; however, the PDF of echo envelope from a fibrotic liver deviates from the Rayleigh distribution. To approximate the PDF of liver fibrosis, we have been proposing the multi-Rayleigh model which is a combination of Rayleigh distributions with different variances. By using the multi-Rayleigh model, the ultrasound B-mode image of liver fibrosis could be converted to the quantitative fibrotic probability image. In this study, we applied the fibrotic probability imaging method to clinical data which is classified into F0 (normal liver), F1 (early-stage hepatitis), F2 (moderate hepatitis), and F3 (late hepatitis) in accordance with a new Inuyama classification. First, a region of interest (ROI) was automatically selected based on the amplitude distribution property. Then, removal method of nonspeckle signals was used to improve the accuracy of the multi-Rayleigh model approximation. Finally, the ultrasound B-mode image in ROI was converted to the fibrotic probability image based on the multi-Rayleigh model.

Invited Paper

11:35

4aBAb2. Ultrasound and microbubbles induced extravasation in a machine-perfused pig liver. Michalakis A. Averkiou, Christina P. Keravnou, and Ine De Cock (BioEng., Univ. of Washington, William H. Foege Bldg., 3720 15th Ave. NE, Seattle, WA 98195, maverk@uw.edu)

Ultrasound driven microbubbles interact with capillaries and cells and have been proposed to be beneficial for localized drug delivery and uptake. Some of the mechanisms involved is transient cell membrane permeability alteration and vessel poration which increase particle extravasation. Ex vivo machine perfusion of human-sized organs is a technique that provides an ideal environment for preclinical investigations with high physiological relevance not possible with *in vitro* experiments. In this work, ex vivo machine-perfused pig livers combined with an image-guided therapy system were used to investigate extravasation induced by ultrasound driven microbubbles. Local microvascular flow changes (measured by contrast enhanced ultrasound) and leakage of Evans blue dye in liver parenchyma were used to assess the degree of extravasation. 1—4 MPa peak negative pressure and 20—1000 cycles were considered. Liver areas that were exposed to long pulses (1000 cycles) and peak negative pressure >2.5 MPa showed a detectable perfusion change. Two fold increase in Evans blue concentration was observed at areas treated with peak negative pressure above 1 MPa and 500 cycles, indicating that extravasation still occurs in lower pressures where flow changes cannot be detected with contrast enhanced ultrasound quantification.

Contributed Paper

11:50

4aBAb3. Enhanced high rate shockwave lithotripsy stone comminution in an *in-vivo* porcine model using acoustic bubble coalescence. Hedieh Alavi Tamaddoni, William W. Roberts (Univ. of Michigan, 2131 Carl A. Gerstacker Bldg., 2200 Bonisteel Blvd., Ann Arbor, MI 48109, alavi@umich.edu), Alexander P. Duryea (HistoSonics, Inc., Ann Arbor, MI), and Timothy L. Hall (Univ. of Michigan, Ann Arbor, MI)

While cavitation on the surface of urinary stones has been shown to aid stone fragmentation, bubbles along the propagation path away from the stone may attenuate shockwaves. We have shown applying low amplitude acoustic waves after each shockwave can force bubble coalescence enhancing lithotripsy efficacy *in vitro*. For the present study, we demonstrate the feasibility of applying acoustic bubble coalescence (ABC) *in vivo*. Artificial model stones were percutaneously implanted in the right kidney of ten 50 kg female pigs and treated with 2500 shockwaves at 120 SW/min with or without ABC. ABC sequences consisted 1 MPa tone bursts at 500 kHz for a total duration of 16 ms. After treatment, kidneys were removed and dissected. Remaining fragments in the kidney were retrieved and filtered for size. Comparing size distributions, a significant improvement was observed when ABC was used. Without ABC, only 25% of the remnant mass of fragments was less than 2 mm. With ABC, 75% of the mass was less than 2 mm. These results suggest ABC can reduce the shielding of cavitation bubbles resulting in a more efficient SWL treatment.

THURSDAY MORNING, 1 DECEMBER 2016 — SOUTH PACIFIC 4, 8:00 A.M. TO 12:00 NOON

Session 4aEA

Engineering Acoustics: Simulation and Analysis

Colin J. Novak, Cochair
Mechanical, Automotive and Materials Engineering, Department of Mechanical Engineering, University of Windsor, 401 Sunset Ave., Windsor, ON N9B 3P4, Canada

Shouichi Takane, Cochair
Department of Electronics and Information Systems, Akita Prefectural University, 84-4 Ebinokuchi, Tsuchiya, Yurihonjo 015-0055, Japan

Contributed Papers

8:00

4aEA1. Integrated simulation model for prediction of acoustic environment of launch vehicle. Seoryong Park, Kiseop Yoon, Jeongwoo Ko, HanAhChim Choung, Seokjong Jang (Aerosp. Eng., Seoul National Univ., Republic of Korea, 1, Gwanak-ro, Gwanak-gu, Seoul, Korea, Seoul 151-744, South Korea, tjfyd11@snu.ac.kr), and Soogab Lee (Ctr. for Environ. Noise and Vib. Res., Eng. Res. Inst. College of Eng., Seoul National Univ., Seoul, South Korea)

During a lift-off of a launch vehicle, a large amount of acoustic wave is generated by jet plumes. Acoustic waves lead to pressure fluctuations on the surface of the vehicle in the form of acoustic loads and these are transmitted to the internal payload. In order to evaluate the effects of acoustic loads, we developed the integrated simulation model. The model is mainly divided into two parts, prediction and reduction parts. First, the external acoustic loads are predicted empirically based on the Distributing Source Method-II (DSM-II) of NASA SP-8072. We developed the improved method able to consider the additional effects of surrounding structures. We perform a next prediction process for the internal acoustic environment by using Statistical Energy Analysis (SEA). For the Korean Sounding Rocket-III (KSR-III), we predict the internal acoustic environment and evaluate the stability of the payload in the fairing. After the prediction process, we perform the noise reduction simulation. With an active noise control (ANC) simulation, we could confirm the possibility and reliability of the noise reduction system.

8:15

4aEA2. Micro load theory of surface acoustic wave gravimetric sensors. Li-Feng Ge (School of Elec. Eng. and Automation, Anhui Univ., 111 Jiulong Rd., Hefei 230601, China, lf_ge@hotmail.com)

Surface acoustic wave (SAW) gravimetric sensors have been widely applied in the field of chemical and biological detections for decades. However, it is noticed that the operating mechanism of this type of sensors is not understood well. A perturbation theory commonly adopted has been used to predict the mass sensitivity of the SAW sensors, but, as indicated also, "there is a significant and systematic quantitative discrepancy" between the theoretical and experimental results [Wohltjen, *et al.* IEEE UFFC, 1987]. In this paper, a micro load theory is presented. The coating film with biochemical material adsorbed is not regarded as a perturbation source, but a micro mechanical load. Then, it is revealed that the fractional frequency shift caused by the load is minus and half of the fractional area density change, where the unloaded area density is a product of the substrate material's density and SAW wavelength. Also, it is interesting to note that the expression is the same with that for Lamb wave sensors if SAW wavelength is replaced by the plate thickness [Li-Feng Ge, POMA, ASA, 2009], which shows a general character of wave-based gravimetric sensors. [The work was supported by NSFC, Grant Nos. 60374044 and 60774053].

8:30

4aEA3. Development of a high presence virtual reality system by binaural sound field rendering with head mount display. Kyosuke Sugiura, Shingo Iino, and Takao Tsuchiya (Sci. and Eng., Doshisha Univ., 1-3 Tatara-Miyakodani, Kyotanabe, Kyoto 610-0394, Japan, dup0154@mail4.doshisha.ac.jp)

In this study, a high presence virtual reality (VR) system which integrates head mount display (HMD) and the binaural sound field rendering is developed. To realize a high presence VR system, it is necessary to provide interactive visual and sound environments in real-time. Recent VR systems can provide interactive visual environment by HMD; however, interactive sound environment is not enough, because enormous binaural room impulse responses (BRIRs) are required to obtain HRTFs at various room positions. So, we calculate BRIRs numerically using the CE-FDTD method with the GPU cluster system. For BRIR calculation, a room model including dummy-head scanned by 3D scanner is developed. In this system, it is possible to render binaural-sound interactively by convolving calculated BRIRs. The developed VR system consists of visual and sound part which is respectively processed by Unity and MAX. Position and head rotation of listener are obtained from tracking sensor by Unity, and the data are sent to MAX by OSC. MAX convolves calculated BRIRs changing by cross-fade. Listener can move their head in virtual room, and can feel sound field differences. It is confirmed that the system can produce the high presence virtual room.

8:45

4aEA4. Three-dimensional numerical simulation of parametric speaker by compact explicit finite-difference time-domain method. Takao Tsuchiya (Dept. of Information Systems Design, Doshisha Univ., 1-3 Tatara-Miyakodani, Kytanabe City, Kyoto 610-0321, Japan, ttsuchiy@mail.doshisha.ac.jp) and Hideyuki Nomura (Commun. Eng. and Informatics, Univ. of Electro-Communications, Chofu, Tokyo, Japan)

In this study, the three-dimensional numerical simulation of a parametric speaker is carried out. To simulate the secondary sound field of difference frequency generated by the high intense ultrasonic wave, the Westervelt equation is numerically solved by the compact explicit finite-difference time-domain (CE-FDTD) method which is a high-accuracy version of the standard FDTD method. In the simulation, the interpolated wideband (IWB) scheme that is one of the most accurate scheme of the CE-FDTD method and whose cut-off frequency is in agreement with the Nyquist frequency, is chosen to reduce computer resources. The IWB scheme is then implemented on a graphics processing cluster system. Three-dimensional numerical models whose size is 1.5 m x 1.5 m x 8 m are developed with cell sizes of 1 mm. Some demonstrations are made for the nonlinear sound wave propagation. The propagation characteristic and the directivity of the parametric speaker are obtained with reasonable accuracy.

9:00

4aEA5. A novel numerical simulation of acoustic Doppler effect using CIP-MOC method. Takuro Sonobe, Kan Okubo, Norio Tagawa (Graduate School of System Design, Tokyo Metropolitan Univ., 6-6 Asahigaoka, Hino, Tokyo 191-0065, Japan, sonobe-takuro@ed.tmu.ac.jp), and Takao Tsuchiya (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan)

Utilizing the Doppler effect (the Doppler frequency shift) is an effective method to detect the speed of the moving object; in particular, the measurement of the acoustic Doppler effect might be an important for recognition of the surrounding environment in the automatic operation and robot technology. To date, many researchers investigated acoustic numerical simulation methods. However, a numerical simulation considering acoustic Doppler effect has not been sufficiently developed yet. In this study, we employ the constrained interpolation profile (CIP) method, one of the methods of characteristics (MOC). A noticeable feature of the CIP method is that it allows analyses not only of sound fields on grid points, but also analyses of their spatial derivatives on grid points, whereas the CIP-MOC method adopts the co-located grid system. That is, sound field components (air pressure and vertical velocity) and their derivatives are located on the same grid, so that we can model accurate interface between different media. This feature is valuable for stable and low-dispersive computation of acoustic Doppler effect. In present study, we first perform some numerical simulations by the CIP-MOC method for sound propagation involving the Doppler effect and then demonstrate the effectiveness of this scheme.

9:15

4aEA6. Comparative study of a low frequency emergency siren to traditional siren technology. Frank Angione, Colin Novak, Chris Imeson, Ashley Lehman, Ben Merwin, Tom Pagliarella, Nikolina Samardzic, Peter D'Angela, and Helen Ule (Univ. of Windsor, 401 Sunset Ave., Windsor, ON, Canada, novak1@uwindsor.ca)

This study measures and compares the acoustic characteristics of a traditional electronic emergency siren and an innovative low frequency Rumbler siren technology. The Rumbler siren's low frequency emissions can travel further and have a greater ability to penetrate and induce structure-bone excitation in nearby vehicle cabins compared to other siren technology. The result is a better ability to warn both nearby vehicles and pedestrians thus lessening the potential of emergency vehicle collisions. The siren technologies were evaluated using three measurement scenarios to determine acoustic localization characteristics, drive-by effectiveness, and the ability to overcome the problem shadowing phenomenon typical at congested intersections. A comparative analysis of the acoustical characteristics of the standalone siren and the addition of The Rumbler system provided insight in regards to the relative effectiveness of each siren mode. The siren system equipped with The Rumbler technology gave a noticeable increase in sound pressure level for each of the three test scenarios. Through analysis of the measured data, the overall performance of the emergency siren system with the Rumbler technology was shown to be a more effective emergency notification device compared to the standalone electronic siren.

9:30

4aEA7. Study of acoustic localization of an emergency siren. Frank Angione, Colin Novak, Peter D'Angela, and Helen Ule (Univ. of Windsor, 2958 Conservation, Windsor, ON n8w5j3, Canada, angionef@uwindsor.ca)

Having the ability to adequately detect the direction of an approaching emergency siren is critical to the effectiveness of the emergency system. Having this ability allows both pedestrians and drivers of nearby vehicles to more quickly and safely react to an approaching emergency vehicle. This study considers a typical electronic siren system that is currently being used by the Windsor Fire & Rescue Services Department. This siren has two fundamental settings: the standard siren signal and the air horn mode, which is typically used when the emergency vehicle is approaching roadway intersections, as these pose the most danger to occupants of both emergency vehicle and general public. Siren and air horn signals were recorded at specific distances from the driver's position at 45° radial increments. From these, the recorded signals and sound pressure levels measured inside the cabin of the vehicle at the various approach angles were used to prepare a subjective jury evaluation to determine the localization characteristics under simulated roadway intersections conditions. Using the outcomes from this study, valuable knowledge was learned which can be applied to future improvements to enhance the localization characteristics of emergency siren systems.

9:45–10:00 Break

10:00

4aEA8. Analysis of the electrostatic ultrasonic airborne actuator using the finite-difference time-domain method. Ryosuke Igai, Takefumi Shibamata, and Minoru K. Kurosawa (Dept. of Elec. and Electron. Eng., Tokyo Inst. of Technol., 4259 Nagatsuta-cho, Midori-ku, Kanagawa, Yokohama 226-8503, Japan, igai.r.aa@m.titech.ac.jp)

We have investigated the structure of an electrostatic airborne ultrasonic actuator by using the FDTD method. The actuator structure has low-mass pleat-fold diaphragms. The pleat designed 10 mm deep and 2 mm wide, vibrate at a frequency range of 10 kHz to 200 kHz. The diaphragms of the pleat vibrate and carry out high power and wide directivity ultrasonic sound waves from a narrow output opening end. We have found the correlation between the depth of the pleat and the wavelength that the sound pressure

level had peak and dip depending on the frequency. Using numerical simulation by the FDTD method, the interference in the pleat animated and described visibly. Instead of the plane wave source that caused the interference of the sound source in the pleats, progressive wave sources have mitigated interference and gained higher outputs. In the FDTD simulation, progressive wave source at 200 kHz generated 73 dBSPL and about +30 dBSPL higher compared to the plane wave when the vibration amplitude and velocity were 33 micron and 41 mm/s at the diaphragm surface. The particle velocity at output point compared to the sound source gained 4.9 times faster at 20 kHz and 7.3 times faster at 200 kHz.

10:15

4aEA9. Some investigations on properties of spatial principal components analysis of individual head-related transfer functions. Shouichi Takane (Dept. of Electronics and Information Systems, Akita Prefectural Univ., 84-4 Ebinokuchi, Tsuchiya, Yurihonjo, Akita 015-0055, Japan, takane@akita-pu.ac.jp)

Analysis for compact representation of spatial change of Head-Related Transfer Functions (HRTFs) based on principal components analysis (PCA), called Spatial PCA (SPCA), is investigated in this report. Although the SPCA of HRTFs has been researched for about 30 years, there exist some questions left unclear. In this report, the author tries to answer the following two questions: (1) how much amount of data (number of subjects and/or directions) is enough for generation of principal components, and (2) which domain (impulse response, frequency spectrum, etc.) is the best for compact representation of individual HRTFs based on the SPCA. As for (1), the amount of data may be relatively small as a result of the preliminary investigation, meaning that the HRTFs of a certain subject can be reconstructed by using the SPCA with the HRTFs of the other subjects of relatively small numbers. As for (2), the author concluded that the complex frequency spectrum brings about the most compact representation as the results of the SPCA of a single Head-and-Torso Simulator (HATS) [S. Takane, Appl. Acoust. **101**, 64-77(2016)]. In this report, whether the same results are obtained or not is examined via comparison of the reconstruction accuracy among the individual subjects. [A part of this research was supported by JSPS KAKENHI (25330202).]

10:30

4aEA10. Numerical simulations of detecting perforated slim hole length in formation with an acoustic method. Shubo Yang, Xiaohua Che, and Wenxiao Qiao (China Univ. of Petroleum, No. 18, Fuxue Rd., Changping District, Beijing 102249, China, yangshubo-cup@foxmail.com)

Water-jet deep penetrating perforation, which utilizes high pressure water-jet to penetrate reservoirs, is a new well interference technology for increasing deliverability. The length of water-jet hole is a key parameter for evaluating water-jet effect. We simulated and compared acoustic fields both inside and outside water-jet hole in different conditions with the time-domain finite difference method. The water-jet hole in the modeling is 3 cm in diameter and 10 m in length. Based on the simulation results, we analyzed the feasibility to detect the length of water-jet hole mainly by adopting low frequency source. Simulation results show that the high frequency source can excite multiple waves inside the water-jet hole, including refraction P and S waves, pseudo-Rayleigh waves, and Stoneley waves. Thus, the signals received in borehole are complex. On the contrary, the low frequency source excites few modes inside the water-jet hole. Specifically, there are only Stoneley waves when the source frequency is lower than 1 kHz. The Stoneley wave velocity is approximately equal to the P wave velocity of the fluid. As a result, we can utilize the arrival time and the velocity of Stoneley waves to calculate the length of water-jet hole.

10:45

4aEA11. Multi-dimensional numerical simulation of sound propagation with background flow using constrained interpolation profile method. Akihiro Fukuda, Kan Okubo (Graduate School of System Design, Tokyo Metropolitan Univ., 6-6 Asahigaoka, HIno, Tokyo 191-0065, Japan, fukuda-akihiro@ed.tmu.ac.jp), Takuya Oshima (Niigata Univ., Niigata, Niigata, Japan), Takao Tsuchiya (Doshisha Univ., Kyotanabe, Kyoto, Japan), and Masashi Kanamori (JAXA, Chofu, Tokyo, Japan)

To date, numerical simulation for sound wave propagation has been widely investigated as a result of advances in computer technology. For acoustic imaging and prediction, the development of accurate and effective analysis methods in time domain has become an important technical issue. The constrained interpolation profile (CIP) method, a novel low-dispersive numerical scheme, is a type of method of characteristics (MOC) employing a co-located grid system. In our past study, we have applied the CIP method to solving the linearized sound propagation equation. MOC is the calculation method which transforms governing equations of acoustic field into advection equations to solve the acoustic fields. Therefore, the method should be suitable for solving sound propagation considering the background flow effect, e.g., prediction of sound noise environment generated from a propulsion system of aircrafts or rockets. In the study, we proposed and examined the calculation method of sound propagation in an arbitrary background flow using CIP method. We made a numerical comparison between proposed method and the linearized Euler equations (LEE) method. The presentation has clarified that the CIP method is more accurate and effective for solving sound propagation in a strong and/or heterogeneous background flow.

11:00

4aEA12. Numerical study of mean flow effect on sound propagation through an aerosol. Ersen Arslan, Yusuf Özyöruk, and Mehmet Caliskan (Middle East Tech. Univ., Dept. of Mech. Eng., G-308, Ankara 06800, Turkey, caliskan@metu.edu.tr)

Previous studies on the sound attenuation and dispersion in an aerosol consisting of air and water droplets have been extended to accommodate mean flow effects. A numerical methodology based on solution of one dimensional, linearized Euler equations in the frequency domain has been developed. The interactions between the fluid and particle phases have been considered to be through Stokesian drag force. First, the verification of the numerical code has been shown for the zero mean flow case with comparative evaluation of results available in literature. Next, the attenuation and dispersion results of both upstream and downstream flows have been presented. The comparison of results suggests that significant deviations exist for attenuation values between the stationary and the mean flow cases at the same frequencies. Keywords: Suspension Acoustics, Finite Difference in Frequency Domain (FDFD), Mean Flow Effect

11:15

4aEA13. Thévenin acoustics. Randall P. Williams and Neal A. Hall (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 10100 Burnet Rd., Bldg. 160, Rm 1.108, Austin, TX 78758, randy.williams@utexas.edu)

Thévenin's theorem can be applied to the analyses of acoustic systems, without limitation to systems that have been reduced to analogous circuit models. The method is particularly convenient in the analysis of acoustic scattering when the scattering object is mobile. In this presentation, the method is illustrated through an alternative derivation of the well-known "mass law" for transmission through a partition and is also applied to the case of acoustic scattering from a rigid, mobile cylinder of arbitrary size in an ideal plane progressive wave. Differences between the conventional solution approach for such problems and the Thévenin-inspired method are discussed, along with the potential benefits of taking such an approach for the simplification of other problems in physical acoustics.

11:30

4aEA14. Evaluation of dependence on calculation parameters in interface settings for finite-difference time-domain analyses of acoustic fields. Chihiro Ogawa, Kan Okubo (Graduate School of System Design, Tokyo Metropolitan Univ., 6-6 Asahigaoka, Rm. 4-312, Hino-shi, Tokyo 191-0065, Japan, ogawa-chihiro@ed.tmu.ac.jp), and Takao Tsuchiya (Faculty of Sci. and Eng. Dept. of Information Systems Design, Doshisha Univ. Graduate School, Kyotanabe-shi, Kyoto-shi, Japan)

To date, numerical analysis for sound wave propagation in time domain has been investigated widely as a result of advances in computer technology. The finite difference time domain (FDTD) methods are very widely used for time domain numerical analysis. The following is a family of FDTD method, for example, the standard FDTD method based on Yee's algorithm, wave equation FDTD (WE-FDTD) method, the FDTD(2,4) method, and so on. The FDTD and WE-FDTD methods cause numerical dispersion error due to using second order finite difference (FD) approximation. To overcome this problem, the FDTD(2,4) method using higher order spatial FDs have been proposed. On the other hand, the settings of the interface between different media are important issue to solve acoustic wave propagation in non-uniform media. In this study, we examine the dependence on calculation parameters in the settings of interface for some FDTD methods. The present study shows dependence on the CFL number and ratio of sound velocity and impedance between the boundaries. We demonstrate that accuracy of the interface between different media strongly depend on calculation parameters and the ratio of sound velocity. Moreover, the proposal settings of the interface for the FDTD(2,4) methods are more accurate.

11:45

4aEA15. Localization of acoustic source in windy environment using ray tracing. Yeong-Ju Go, Donghun Choi, Jaehyung Lee, and Jong-Soo Choi (Chungnam National Univ., 99 Daehak-ro, Yuseong-gu, Daejeon 34134, South Korea, yjgo@cnu.ac.kr)

Position estimation of sound source using ray tracing method is introduced based on wind field measurement. TDOA (time difference of arrival) estimate error occurs during outdoor sound measurement due to the atmospheric condition such as wind, temperature, humidity and so on. Such condition can possibly change the trajectories of rays. It produces erroneous propagation time information accordingly. In this study, simplex method is used to estimate the source position using ray tracing. The basic concept is to find the estimation point which generates smallest mean square error of propagation distances or times from the candidate points. At first the estimation area is set around the first estimate which is calculated from the TDOA estimation. The simplex method is then applied to each point in the estimation area. Nelder-Mead simplex method is used to find the best position that produces the minimum error in travelling distances. In order to apply the ray tracing, the wind distribution is interpolated based on the wind measurements around the boundary. Simulation and experiments are conducted to verify the performance of the algorithm.

THURSDAY MORNING, 1 DECEMBER 2016 — CORAL 5, 8:00 A.M. TO 10:25 A.M.

Session 4aID

Interdisciplinary: Workshop for Publishing Excellence in JASA

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

James Lynch, Cochair
Woods Hole Oceanographic, MS # 11, Bigelow 203, Woods Hole Oceanographic, Woods Hole, MA 02543

Chair's Introduction—8:00

Invited Papers

8:05

4aID1. Choosing venues (including e-mail, internet posting, and archival journals) for the dissemination, discussion, and publication of research. Allan D. Pierce (Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net)

This paper is intended for researchers desiring that their papers get the attention of relevant research community members and will remain widely accessible for the indefinite future. It is suggested that authors send preprints of their works as electronic attachments to all those colleagues they suspect of having possible interest in the work. They should also consider the possibility of posting their preprints on personal and institutional web sites and on e-print servers such as arXiv. They may also want to consider establishing accounts on social networking sites such as ResearchGate so they can make their publications available to all other account holders (worldwide) who explicitly choose (and are enticed by ResearchGate) to follow their work. Before selecting a journal for submission of any paper, they should familiarize themselves with the terms of the copyright agreement and decide whether the scope of authors' rights is sufficient for their desires. Various factors important for the selection of a journal are discussed: relevance of previously published papers to the author's field of research, its likelihood of perpetual existence, the familiarity of the editors with the field, the quality reputation of the past published papers, and the timeliness of the processing of submitted articles.

8:25

4aID2. The publication process as seen from various angles. James Lynch (Woods Hole Oceanographic, MS # 11, Bigelow 203, Woods Hole Oceanographic, Woods Hole, MA 02543, jlynch@whoi.edu)

The process of publishing a paper in JASA involves many people and several process components. The authors, reviewers, Associate Editors, Publication Office personnel, and our AIPP publishers all play their parts and interact to produce any paper that is eventually published. In this talk, the various components of the publication process will be examined, making an effort to present the tasks and viewpoints of the various participants. The merits and deficiencies of various parts of the publication process will also be discussed, with the goal of both promoting understanding of the system and also improving it.

8:45

4aID3. Preparing and submitting manuscripts to the Journal of the Acoustical Society of America—Express Letters. Charles C. Church (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, cchurch@olemiss.edu)

The general process for preparing a manuscript for the Journal of the Acoustical Society of America—Express Letters (JASA-EL) is the same as for any other scientific publication. First, have something new and interesting to say to your colleagues. Second, say it clearly, without ambiguity, and with a minimum of unnecessary literary flourish. Finally, and most importantly for JASA-EL, be brief. Failure to follow any of these simple rules may produce lengthy delays in publication. In extreme cases, it may result in rejection of the manuscript even after the review process has been completed! Once a manuscript has been completed, it is very important to follow the formatting requirements for the journal, both for the length and the appearance of the text and tables of the manuscript itself but also for the file types accepted by our publisher, AIP (American Institute of Physics) Publishing. The submission procedure is relatively simple. For example, neither a cover letter nor a list of suggested reviewers is currently required (although this may change in the future!). However, there are a few characteristics of the process that authors should consider before beginning. These and other areas that are frequently sources of difficulty will be discussed.

9:05

4aID4. Instructions for prospective authors to submit acceptable manuscripts to the Journal of the Acoustical Society of America. Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu)

It is a common perception that the Journal of Acoustical Society of America (JASA) is considered as one of leading journals which has been serving the acoustics community for over 85 years. The JASA encourages authors to submit papers for JASA publication and at the same time practices a rigorous peer-review process to meet its aspirations for a higher degree of publishing excellence. For prospective authors a better understanding of JASA's peer-review process is critically important. To increase the chance for submission acceptance, this talk will shed light to its peer-review process of the JASA, draw audience's attention to some detailed, updated guidance and instructions regularly published by the JASA editorial board. As a frequent reviewer, an Associate Editor, and particularly a non-native author of the JASA papers, this author will share his own experience to analyze possible reasons for successful and unsuccessful publication effort using some disguised examples.

9:25

4aID5. Best practices for publishing in scientific journals. Christy K. Holland (Internal Medicine, Div. of Cardiovascular Diseases and Biomedical Eng. Program, Univ. of Cincinnati, Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

A string of impressive publications can be the lifeline for promotion and success for an academic. Once excellent data is produced from your innovative research program, how do you communicate the results as clearly as possible? As the editor in chief of Ultrasound in Medicine and Biology, I have witnessed first hand a range of successful strategies to get papers published. Making sure that your manuscript is aimed at the appropriate journal at the outset can save a great deal of effort and can get your work into print faster. Editors and Associate Editors seek advice from reviewers who are knowledgeable in the general subject of the paper, and the reviewers give opinions on various aspects of the work, including whether the work is original and correct. The editor and reviewers who examine each manuscript are the authors' peers: persons with comparable standing in the same research field as the authors themselves. The composition of a concise cover letter, the creation of an effective response to the reviewers' critiques, and strategies for polite communication will be outlined and discussed. The good news is that editors want to publish your paper and appreciate individuals who create the scientific content that makes a journal thrive.

9:45–10:25 Panel Discussion

Session 4aMUa

Musical Acoustics: Flute Acoustics Including All "Air-Jet" Instruments

Naoto Wakatsuki, Cochair
University of Tsukuba, 1-1-1 Tennodai, Tsukuba 305-8573, Japan

Nicholas Giordano, Cochair
Physics, Auburn University, College of Sciences and Mathematics, Auburn, AL 36849

Chair's Introduction—8:00

Invited Papers

8:05

4aMUa1. How well do lumped-element models describe acoustic amplification in the recorder? John C. Price (Phys., Univ. of Colorado, 390 UCB, Boulder, CO 80309, john.price@colorado.edu)

Nearly 50 years have passed since Cremer and Ising [Acustica **19**, 142-153 (1967)] established lumped-element feedback models as the dominate paradigm for understanding flute-family musical instruments. Modern versions of these models include one feedback path through the resonant acoustic flow in the pipe, and a second path that accounts for edge-tone phenomena. They also incorporate results of later experimental and theoretical work on jet formation, jet deflection, and growth of instabilities along the jet. In this work, we test a lumped model of the recorder air-jet amplifier by replacing the recorder body with a waveguide reflectometer. When the mean flow from the air-jet into the waveguide is not blocked, the air-jet amplifier is unstable to edge-tone oscillations. When it is blocked, the air-jet is deflected somewhat outward, gain is reduced, and the system becomes stable. It is then possible to measure the reflection coefficient of the air-jet amplifier versus blowing pressure and acoustic frequency under linear response conditions, avoiding the complication of gain saturation. The results provide a revealing test of lumped flute drive models under simple conditions and with few unknown parameters. The strengths and weaknesses of modern flute drive models are discussed.

8:25

4aMUa2. Direct aeroacoustic simulations and measurements of flow and acoustic fields around a recorder with tone holes. Hiroshi Yokoyama (Mech. Eng., Toyohashi Univ. of Technol., 1-1,Hibarigaoka,Tempaku-cho, Toyohashi, Aichi 4418580, Japan, h-yokoyama@me.tut.ac.jp), Akira Miki (Res. & Development Div., Yamaha Corp., Shizuoka, Japan), Ryoma Hamasuna (Mech. Eng., Toyohashi Univ. of Technol., Toyohashi, Aichi, Japan), Hirofumi Onitsuka (Res. & Development Div., Yamaha Corp., Shizuoka, Japan), and Akiyoshi Iida (Mech. Eng., Toyohashi Univ. of Technol., Toyohashi, Aichi, Japan)

Direct aeroacoustic simulations and experiments were performed regarding the flow and acoustic fields around a recorder, which is one of air-reed instruments. The present simulations reproduced the jet oscillations and acoustic radiation around the recorder with tone holes. The effects of conditions of the tone holes and the jet velocity on the flow and acoustic fields are discussed.

8:45

4aMUa3. Acoustic energy generation of "air-jet" instruments: Energy transfer between jet oscillation and acoustic field. Kin'ya Takahashi (The Phys. Labs., Kyushu Inst. of Technol., Kawazu 680-4, Iizuka, Fukuoka 820-8502, Japan, takahasi@mse.kyutech.ac.jp), Sho Iwagami (Mech. Information Sci., Kyushu Inst. of Technol., Iizuka, Fukuoka, Japan), Taizo Kobayashi (Faculty of Fukuoka Medical Technol., Teikyo Univ., Omuta, Fukuoka, Japan), and Toshiya Takami (Comput. Sci. and Intelligent Systems, Oita Univ., Oita, Oita, Japan)

In this talk, we discuss how to estimate the acoustic energy generation of "air-jet" instruments with numerical simulation. To attack this problem, we use Howe's energy corollary, with which we can estimate energy transfer between unsteady flow, i.e., oscillating jet and acoustic field. To calculate Howe's formula, we need solenoidal velocity of the flow and its vorticity together with acoustic particle velocity separated from the whole velocity of compressible fluid. Recently, a method, which allows us to approximately calculate Howe's formula, was developed in experiments by Bamberger and Yoshikawa *et al.*, and it can be applied for the numerical calculation. We apply the method for the numerical calculation of a flue organ pipe model. We also introduce a toy model of the oscillating jet to investigate the mechanism of sound generation from the oscillating jet in detail. The acoustic energy is mainly generated in the downstream of the oscillating jet near the edge of the mouth opening, but it is consumed in the upstream near the flue exit to synchronize the jet motion with it. Our results are in good agreement with the experimental result by Yoshikawa *et al.* as well as Howe's theoretical prediction.

9:05

4aMUa4. Simulation studies of the flute. Nicholas Giordano (Phys., Auburn Univ., College of Sci. and Mathematics, Auburn, AL 36849, njg0003@auburn.edu)

Simulations of the transverse flute using direct numerical solutions of the Navier-Stokes equations are described. Constructing a numerical model of the instrument itself is straightforward, but making a semi-realistic model of the player's lips is more challenging. One solution to this problem is described and used to explore the kinds of tones that are possible. Conditions are found for which the model produces a well-behaved musical tone for a range of blowing speeds. When the blowing speed is increased beyond this range the behavior becomes complex and is appears to be chaotic. An analysis of this behavior in terms of the sensitivity to initial conditions and a route to chaos involving Hopf bifurcations is described. The relevance of these results to real instruments played by real players is also discussed. [Research supported by NSF grant PHY1513273.]

9:25

4aMUa5. Study on the principle of sound resonance in human whistling using physical models of a human vocal tract. Mikio Mori (Univ. of Fukui, 3-9-1 Bunkyo, Fukui City, Fukui 910-8507, Japan, mikoi@u-fukui.ac.jp)

The principle behind sound production in human whistling is relatively unknown. A good understanding of this principle would be beneficial to both trainer and trainee. Rayleigh identified whistling frequency as being determined by the mouth cavity and pointed out that earlier ideas that relate the sound-producing mechanism to the vibration of the lips are inaccurate. Wilson *et al.* reported that the human whistling resonant frequency is close to the Helmholtz resonant frequency, and through some physical-model-based experiments, they determined that the resonator can be excited by a flow through the smooth-edged orifices bounding the resonant cavity. Thus, the principles of whistling in terms of articulation have been reported to be based on the Helmholtz resonance. However, some whistlers can produce high-pitched sounds by blowing harder without changing the capacity of the resonance chamber, which is similar to a high-pitched sound produced by a wind instrument by air-column resonance. This work studies the principle of sound production during human whistling using physical models of a human vocal tract, which is obtained from an X-ray computed tomography image during whistling. We have demonstrated that the principle of resonance in human whistling includes not only the Helmholtz resonance but also an air-column resonance.

Contributed Papers

9:45

4aMUa6. Numerical study on acoustic intensity of edge tone with change of jet velocity. Iwagami Sho (Mech. Information Sci. and Technol., Kyushu Inst. of Technol., Kawazu 680-4, Iizuka, Fukuoka 820-8502, Japan, iwagami@chaos.mse.kyutech.ac.jp), Taizo Kobayashi (Faculty of Fukuoka Medical Technol., Teikyo Univ., Omuta, Fukuoka, Japan), Kin'ya Takahashi (Mech. Information Sci. and Technol., Kyushu Inst. of Technol., Iizuka, Fukuoka, Japan), and Yuji Hattori (Inst. of Fluid Sci., Tohoku Univ., Sendai, Miyagi, Japan)

To understand the sounding mechanism of "air-jet" instruments we numerically study the acoustic intensity of edge tone, i.e., the sound source of "air-jet" instruments, with compressible LES and DNS comparing with experimental results. The edge tone is aerodynamic sound and we need to investigate the sound generation based on Lighthill's theory. According to Lighthill's theory, the acoustic intensity increases the eighth power of fluid velocity in transonic and subsonic ranges, but it may obey different rules in a range less than the subsonic. Indeed, we find with compressible LES that the edge tone has three stages, in each of which a different increasing rate is observed. In particular, the fundamental and second modes obey the second-power law in the second stage. The same result was obtained by an experiment provided by YAMAHA Corporation. To clarify the radiation mechanism of the edge tone, which exhibits the three stage, we have to calculate Lighthill's sound source with high accuracy. So, we will show our effort in this direction: improving the calculation with compressible LES together with DNS, which gives very accurate results at low Reynolds numbers.

10:00

4aMUa7. Numerical study of the influence of the mouth-flue-foot geometry on sounding mechanism of an "air-jet" instrument model. Shuhei Tateishi, Genki Tsutsumi (Mech. Information Sci. and Technol., Kyushu Inst. of Technol., Kawazu 680-4, Iizuka, Fukuoka 820-8502, Japan, shuuhei.tateishi@gmail.com), Taizo Kobayashi (Facalty of Fukuoka Medical Technol., Teikyo Univ., Omuta, Fukuoka, Japan), Toshiya Takami (Comput. Sci. and Intelligent Systems, Oita Univ., Oita, Oita, Japan), and Kin'ya Takahashi (Mech. Information Sci. and Technol., Kyushu Inst. of Technol., Fukuoka, Fukuoka, Japan)

The geometry of flue and foot influences the stability of jet oscillation, which products aerodynamic sound as the sound source of "air-jet instruments. Thus, we numerically study the influence of mouth-flue-foot geometry on the stability of jet oscillation with compressible LES. We introduce a 2D flue organ pipe model terminated with a closed end. We investigate the influence of the length of the flue, the existence of chamfers at the flue exit and the volume of the foot. As a result, eliminating the chamfer makes the jet oscillation unstable, while adding the chamfer stabilizes the jet oscillation and sound generation. A long flue makes the jet motion stable and robust, so that it spends a long time to reach a stable oscillation and cannot response quickly to the change of the air supply. Thus a relatively short flue with the chamfers is suitable for a performance. This result qualitatively agrees with the experimental result reported by Segoufin *et al.*. We also found that the existence of foot stabilizes the sound oscillation in the pipe: the pressure oscillation in the foot synchronizes with that in the pipe with a phase delay, but eliminating the foot rather makes the sound oscillation unstable.

Session 4aMUb

Musical Acoustics: Piano Acoustics and Playing Piano I

Isoharu Nishiguchi, Cochair
Information Media, Kanagawa Institute of Technology, 1030, Shimo-ogino, Atsugi 243-0292, Japan

Nicholas Giordano, Cochair
Physics, Auburn University, College of Sciences and Mathematics, Auburn, AL 36849

Chair's Introduction—10:30

Invited Papers

10:35

4aMUb1. Measurement and numerical simulation of two-dimensional vibration of a single piano string. Daisuke Naganuma (Tokyo Int. Univ., 1-13-1, Matobakita, Kawagoe, Saitama 350-1197, Japan, naganuma@tiu.ac.jp), Hideyuki Nomura, and Tomoo Kamakura (The Univ. of Electro-Communications, Chofu, Tokyo, Japan)

A piano string begins to vibrate vertically as soon as a hammer strikes it. After the vertical vibration has begun, the string then starts to rotate owing to the generation of horizontal vibration. The rotation direction changes several seconds later, which suggests that the frequencies of the vertical and horizontal components of the vibration are slightly different. We previously reported an equivalent mechanical circuit model for physically elucidating such two-dimensional movements of the string. In this presentation, we show the verification and validation of the equivalent model by comparing with experiments. The movements of a string in low frequencies were observed using a high-speed camera and a mirror. In order to increase the measurement accuracy of displacement, interpolation technique was applied to the correlation of recorded images. It was noted that the frequency of a partial vibration of the string in the vertical direction is higher or lower than that in the horizontal direction to the soundboard. The rotation directions and the rotation change periods suggest how the soundboard acts on the generation of the partial vibration as mass-dominated or stiffness-dominated control at the string end, and to what extent does the soundboard affect the partial vibration of the string.

10:55

4aMUb2. Simulation of three-dimensional motion of a piano string using detailed and simplified computational models. Isoharu Nishiguchi (Information Media, Kanagawa Inst. of Technol., 1030, Shimo-ogino, Atsugi, Kanagawa 243-0292, Japan, nishiguc@sd.kanagawa-it.ac.jp)

It is well known that the decay curve of the piano sound is steep for the first few seconds and followed by slower decay at longer time. The causes of this double decay are also known: the effect of polarization of a string which is considered to be caused by the motion of the bridge which is connected to the string by two bridge pins and interaction among multiple strings for a note. As for the later, it is explained that multiple strings for a note tend to vibrate out of phase each other when they have slightly different frequencies and this suppresses the energy transferred to the soundboard through the bridge. However, to make quantitative evaluations of these causes, it is indispensable to develop a physical model of the piano in which contributions of each factor of these causes can be simulated with a necessary degree of accuracy. In this study, the finite element method and the mode superposition method are employed to simulate the three-dimensional motion of a piano string in which interaction with a bridge on a soundboard is taken into account. The polarization and the longitudinal vibration of a piano string were measured to assess the simulation.

11:15

4aMUb3. Localization in the piano soundboard. Gautier Lefebvre, Marcel Filoche, and Xavier Boutillon (Physique de la Matière Condensée, École polytechnique, CNRS, Université Paris-Saclay, Laboratoire de Mécanique des Solides, Aile 3, École polytechnique, Palaiseau 91128, France, lefebvre@lms.polytechnique.fr)

The soundboard is the complex plane structure that radiates the piano sound. We focus on its structural vibrations. The arrangement of the stiffeners (parallel bars) displays some disorder (height, separation distance) that has been shown responsible for the localization of the vibrations above a precise frequency (Ege 2013, *JSV*; Chaigne 2013, *JASA*) By means of a finite-element method, we investigate the localization properties of the eigenmodes of a simplified soundboard model. The localization occurs close to the band-edge of the first Brillouin zone, indicating the importance of a slightly disordered structure, similarly to photonic or phononic crystals with defects (John 1987, *PRL*). For structural parameters typical of a piano soundboard, this phenomenon seems to force the soundboard to remain in the subsonic regime in a very large frequency range. Moreover, by the adaptation of the theory of the localization landscape (Filoche 2012 *PNAS)*, we compute a dual landscape (Lyra 2015 *EPL*) for the high-frequency modes of a stiffened orthotropic thin plate. This enables to predict the position of localized eigenmodes. [Work supported by the ANR grant ANR-14-CE07-0014.]

11:35

4aMUb4. Real-time physical model of a grand piano soundboard on field programmable gate array hardware. Florian Pfeifle and Rolf Bader (Univ. of Hamburg, Neue Rabenstrasse 13, Hamburg 20354, Germany, Florian.Pfeifle@uni-hamburg.de)

A modern grand piano consists of over 10,000 singular parts and has a production cycle of approximately one year. This leads to the predicament that researching and changing acoustical features resulting from variable material properties of the instrument cannot be realized in a straightforward manner. The inevitable time delay as well as the variability of parameters makes a controlled research very hard, if not impossible. In this work, a system is presented that enables piano makers to change, simulate as well as auralise acoustical vibrations of a grand piano soundboard in real time using physical modelling methods on field programmable gate array (FPGA) hardware. This specific implementation focusses on synthesizing and simulating the vibrations of a geometrically complete model of a concert grand piano soundboard, including the ribs, the bass, and main bridge, realistic material parameters as well as coupled strings. It can be parametrized and controlled from a software GUI running on a personal computer. This implementation is part of a larger project aiming at modeling a complete grand piano in real-time for research, instrument making as well as purely musical applications.

THURSDAY MORNING, 1 DECEMBER 2016 SOUTH PACIFIC 3, 8:00 A.M. TO 11:50 A.M.

Session 4aNS

Noise: Prediction of Transportation Noises and Other Topics I

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

Shinichi Sakamoto, Cochair
5th Department, Institute of Industrial Science, The University of Tokyo, 4-6-1, Komaba, Meguro-ku, Tokyo 153-8505, Japan

Chair's Introduction—8:00

Invited Papers

8:05

4aNS1. Evaluation of lateral attenuation for aircraft flyover and ground operation noise in airport noise modeling. Naoaki Shinohara (Aviation Environment Res. Ctr., AirportI Environment Improvement Foundation, Ryuuen Bldg., 1-3-1, Shiba-Koen, Minato-ku, Tokyo 150-0011, Japan, n-shinohara@aeif.or.jp), Kazuyuki Hanaka (Narita Int. Airport Fromotion Foundation, Shibayama, Chiba, Japan), and Ichiro Yamada (Aviation Environment Res. Ctr., Airport Environment Improvement Foundation, Minato-ku, Tokyo, Japan)

At the side of flight path, especially on the side of runway, noise level observed greatly changes dependent on the effects of excess ground attenuation and meteorological conditions (in particular, vector wind and temperature gradient). Thus, taking account of appropriate adjustment for lateral attenuation in airport noise modeling leads to an increase in the reliability of predictions. This paper makes a review of equations for lateral attenuation such as SAE/AIR 1751, SAE/AIR 5662, and our equation 1751M, which is a modified equation we proposed based on AIR/1751 and is applicable under various meteorological conditions. It is also described with further issues related to the examining of lateral attenuation for aircraft flyover noise. In Japan, it is obligatory to take into account noise contributions of aircraft ground operations such as APU operation, taxiing, and engine run-up in airport noise modeling. However, we have not yet fully examined whether we can evaluate lateral attenuation of such ground noise similarly to flyover noise. It is not yet solved how to calculate excess ground attenuation under the effects of sound shielding by barriers or embankment. This paper also discusses how to evaluate lateral attenuation for noise due to aircraft ground operations.

8:25

4aNS2. Study of lateral attenuation for aircraft noise modeling based on field measurements near the airport. Kazuyuki Hanaka (Narita Int. Airport Promotion Foundation, 76 Amazutsumi Kayamashinden Shibayama-Town,Chiba, Shibayama, Chiba 289-1601, Japan, hanaka@napf.or.jp), Naoaki Shinohara, and Hisashi Yoshioka (Airport Environment Improvement Foundation, Tokyo, Japan)

It is essential to appropriately evaluate effects of lateral attenuation in predicting aircraft noise at the side of flight path near the airport. About fifteen years ago, we examined the validity of existing equations of lateral attenuation, but noise characteristics of aircraft and engines have changed very much. Especially, in case of horizontal sound propagation over ground such as aircraft rolling on the runway, it is necessary to take into account that meteorological conditions may strongly affect lateral attenuation. Thus, we carried out field measurements of aircraft noise repeatedly on the side of runway at several airports including Narita Airport and examined the actual

situation of lateral attenuation. As a result, it was found that lateral attenuation greatly changes dependent on meteorological conditions and that calculations from existing equations of lateral attenuation did not match measurements of lateral attenuation. Additionally, considering actual situation of land use around the airport, it is not always realistic to assume that the ground surface is flat and uniform as well as acoustically soft. It is sometimes necessary to assume that acoustic property of the ground changes. It is also necessary to evaluate lateral attenuation on conditions that the sound propagates over urbanized city area.

8:45

4aNS3. Evaluating aircraft ground operation noise in airport noise modeling and monitoring in Japan. Ichiro Yamada (Airport Environment Improvement Foundation, Ryuuen Bldg. 1-3-1, Shiba-Koen, Minato-Ku, Tokyo 105-0011, Japan, i-yamada@aeif.or.jp)

The current Japanese noise guideline “Environment Quality Standard for Aircraft Noise” requires considering noise of aircraft ground operation, which we call ground noise here, like APU operation, taxiing, and engine run-up, if necessary. It has inevitably required us not only to take account of ground noise contributions in the Japanese airport noise model JCAB2, but also to include techniques to identify or discriminate ground noise from the ambient noise floor when fulfilling airport noise monitoring. Ground noise sometimes lasts very long and is usually low level compared with flyover noise. First, this paper makes a brief review of the way to evaluate ground noise in airport noise modeling and monitoring in Japan. Second, it discusses further issues to improve the reliability in the evaluation of ground noise under the influences of excess ground attenuation, meteorological condition, sound shielding by obstructions like building structures and embankment, as well as contamination by noise events.

9:05

4aNS4. An aircraft noise prediction model considering meteorological effects on noise propagation developed for air traffic optimization. Hirokazu Ishii, Adriana Andreeva-Mori (Japan Aerosp. Exploration Agency, 6-13-1 Osawa, Mitaka, Tokyo 181-0015, Japan, ishii.hirokazu@jaxa.jp), Takatoshi Yokota, Koichi Makino, and Toshio Matsumoto (Kobayasi Inst. of Physical Res., Tokyo, Japan)

With the growing transportation demand, the volume of air traffic is already approaching and will soon exceed the capacity limit posed by current air traffic control. Japan Aerospace Exploration Agency (JAXA) has conducted a research project called DREAMS, aiming to develop technologies for future air traffic management system. One of the research topics examined by the project is noise abatement flight technology. It optimizes approach paths considering the meteorological effects on noise propagation. Its key element is an aircraft noise prediction model which can take into account the effect of meteorological conditions on noise propagation. The effect is computed by Green’s Function-Parabolic Equation method and verified by field experiment using an elevated sound source. Aircraft noise is measured around approach flight path for overall verification, and the results show the average and the standard deviations of the prediction error are -0.66 dB and 1.55 dB, respectively. The presentation provides an overview of the development and the verification of the noise prediction model, followed by a comparison between noise footprints calculated with JAXA’s model and the Integrated Noise Model (INM). Application of the noise prediction model to descent flight path optimizations for noise abatement is also shown.

9:25

4aNS5. A field survey on sound power level and spectrum of Japanese road vehicles. Miki Yonemura (School of Eng., The Univ. of Tokyo, Komaba 4-6-1, Inst. of Industrial Sci., Ce402, Meguro-ku, Tokyo 153-8505, Japan, m-yone@iis.u-tokyo.ac.jp), Hyojin Lee, and Shinichi Sakamoto (Inst. of Industrial Sci., The Univ. of Tokyo, Meguro-ku, Tokyo, Japan)

For precise prediction and appropriate countermeasure of road traffic noise, to grasp the sound power level and power spectrum of running vehicles is important. In Japan, there exists a road traffic noise prediction model “ASJ RTN-Model 2013,” which provides a simple sound source modeling and simple calculation methods of the noise propagation. Recently, vehicles with good fuel-efficiency such as hybrid vehicles are increasing in the category of passenger car. Such vehicles may have less noise emission than previous gasoline engine vehicles. The authors conducted field surveys to grasp the noise of the current road vehicles especially in the category of passenger cars. As a result of that, in terms of A-weighted sound power levels, which can be expressed as a function of running speed, the difference between gasoline engine vehicles is quantitatively discussed. As for the power spectrum, there is no appreciable difference between gasoline engine vehicles and hybrid vehicles. However, both types of vehicles have more or less different spectral characteristics from ASJ RTN-Model 2013.

Contributed Paper

9:45

4aNS6. Effect of accuracy of digital map data on predicting road traffic noise in Japan. Yasuhiro Hiraguri (Dept. of Civil Eng. and Architecture, National Inst. of Technol., Tokuyama College, Gakuen-dai, Shunan, Yamaguchi 7458585, Japan, hiraguri@tokuyama.ac.jp) and Kazutoshi Fujimoto (Kyushu Univ., Fukuoka, Japan)

Use of GIS for evaluating the Environmental Quality Standards for Noise is established at local authorities in Japan. A digital map on a scale of 1/2,500 is generally used. However, it has not been clarified yet whether the digital map with such a scale has an accuracy to apply for the estimation of environmental noise. A ground plan of buildings on the digital map is a roof plan, because the digital map data are made on a basis of an aerial photograph. It is feared that the noise estimated by using such a map may have some differences in the effects of buildings on the noise propagation from reality. In spite of the fact that the propagation of road traffic noise is highly affected by whether a road is visible from receiving point or not, when the roof plan is used as the floor plan, the road is sometimes visible from the receiving point on the digital map, though it is invisible in practice. The aim of this study is, therefore, to reveal the effect of accuracy of the digital map in GIS on the estimation of road traffic noise for evaluation of the Environmental Quality Standards for Noise.

10:00–10:15 Break

Invited Paper

10:15

4aNS7. Modeling noise from motocross facility with complex terrain. Leisa Nalls and Richard Carman (Wilson Ihrig, 6001 Shellmound, Ste. 400, Emeryville, CA 94608, lnalls@wiai.com)

The authors prepared a study for an existing motocross facility under new ownership. The study was required by the county for a continued use permit. The facility is in a rural setting and will host weekend races and practice activity. The terrain of the motocross track is complex with a hill and flat section with jumps. To conduct the study, an empirical noise model was constructed for future track activity using noise measurements performed at selected locations at the facility's property line during simulated race and practice activity. A model of the future dirt bike noise was created by scaling the measured race and practice activity based on the level of future bike activity anticipated by the owner. The data were scaled using the number of bike rider hours to account for the time riders spend riding, the number of riders per activity, and the number of different activities per day. Long-term noise was measured at the same locations to obtain typical ambient levels during hours with no bike activity. The expected Ldn noise exposure at the property line was obtained by combining the projected race and practice activity noise for different scenarios with the typical ambient noise.

Contributed Papers

10:35

4aNS8. Four methods of estimating population exposure to transportation noise. Enda Murphy and Owen Douglas (Univ. College Dublin, Dublin, Dublin 1, Ireland, enda.murphy@ucd.ie)

This paper assesses four methods for estimating population exposure to road traffic noise within the context of the EU Environmental Noise Directive (END). Employing a case study in central Dublin, Ireland, the methods—MAX, MIN, VBEB/CNOSSOS and AVE—are tested utilising the L_{den} and L_{night} indicators. The study first investigates the extent to which exposure estimates may vary depending on the method utilised while controlling for the noise calculation method. Second, it investigates how estimates of exposure vary depending on the calculation method used; in this case, CRTN and NMPB. The results show that controlling for noise calculation method and employing the same input data, estimations of population exposure differ substantially depending on the exposure method employed. Furthermore, the potential variability in estimated night-time exposure and the potential for under or over estimation of the health effects of environmental noise on a given population when different methods are utilised is clearly demonstrated. The results also show that the method of noise calculation employed has an effect on estimated exposure, particularly for L_{den} measures. Values for L_{night} were found to be very similar regardless of the calculation method employed.

10:50

4aNS9. Sound source identification on automotive muffler surface using single explosion excitation and near-field acoustic holography method. Ryota Matsumoto, Takehiko Seo, Masato Mikami (Mech. Eng., Yamaguchi Univ., 2-16-1 Tokiwadai, Ube, Yamaguchi 755-8611, Japan, v051ve@yamaguchi-u.ac.jp), and Takashi Esaki (Sango Co., Ltd., Miyoshi, Japan)

Recent automobiles tend to have larger interior space, resulting in the smaller space for arranging the muffler. Flat shape mufflers are required for this reason but cause relatively large radiation noise from the surface. In order to improve understanding of radiation noise generation mechanism of flat shape muffler, we tried to identify the noise sources using the near-field acoustic holography (NAH) method on muffler surface excited by a single explosion pulse, which simulates a single pulsation in the exhaust system,. Two types of mufflers were tested with different shapes. Results show that some areas had large vibration velocity at specific frequencies depending on the muffler shape. The flat muffler tends to generate large radiation noise and vibration velocity as compared with a conventional one. In the front and rear outer plates of the muffler, large vibration velocity exists within a certain range of frequency. Within the same frequency range, the flat muffler has more number of the peak of radiation noise as compared with the conventional one.

11:05

4aNS10. Robustness of the extended delayed-X harmonics synthesizer algorithm applied to engine exhaust noise control. Sergej Jukkert and Delf Sachau (Mechatronics, Helmut-Schmidt-Univ. Hamburg, Holstenhofweg 85, Hamburg, Hamburg 22043, Germany, jukkerts@hsu-hh.de)

Exhaust noise of a combustion engine contains several harmonic components of the fundamental frequency that depends on the engine speed. Active noise cancellation (ANC) in the exhaust line of a diesel-electric drive faces serious challenges such as high temperature, high sound pressure levels, standing waves phenomena, a time-varying environment, and slightly varying disturbance frequencies. In the case of a non-acoustic reference sensor, also an offset or a mismatch in the estimation of the disturbance frequencies negatively affects the performance. The extended delayed-X harmonics synthesizer (DXHS) is known from several researches to be appropriate for control of periodic noise with multiple harmonic components. The algorithm is robust against observation noise and fluctuation of the plant under control. In this paper, the extended DXHS is implemented in an ANC-system for a diesel-electric drive and experimentally investigated in laboratory experiments. A modified update rule, a convergence factor regulation, and the pre-filtering of the error signal are implemented to improve the DXHS. Performance and robustness are tested in a mockup of an exhaust line in which realistic fluctuations of temperature and gas flow as well as varying frequencies and amplitudes of the exhaust noise are simulated.

11:20

4aNS11. A study on transient noise-generation characteristics in a diesel engine with rapid premixed-charge compression ignition combustion. Shodai Sagara, Yoshiki Sumida, Takehiko Seo, and Masato Mikami (Mech. Eng., Yamaguchi Univ., 2-16-1 Tokiwadai, ube, Yamaguchi, Ube, Yamaguchi 755-8611, Japan, v014ve@yamaguchi-u.ac.jp)

This study investigates time-frequency characteristics of noise generation in a diesel engine with rapid PCCI (premixed-charge compression ignition) combustion. We increased the pilot injection quantity from a base PCCI condition to simulate rapid PCCI combustion. We measured the in-cylinder pressure and sound pressures and analyzed them using fast Fourier transformation and wavelet transformation. The main cause of the engine noise is the combustion impact on the combustion chamber. We clarified the process that the combustion impact vibrates the engine structure and then lead to sound radiation. We identified main frequencies, whose noise components largely affect the overall level of engine noise. Almost all engine noise components at the main frequencies were found to be strongly affected by the combustion impact. We researched the noise source and transmission path of each noise component by using a sound field visualization device and measuring accelerations on the engine surface.

11:35

4aNS12. Proposal of suspended floor structure for reducing interior noise of railway vehicle. Tomonori Goto and Katsuya Yamamoto (Vehicle Structure Technol. Div., Railway Tech. Res. Institute, 2-8-38, Hikari-cho, Kokubunji-shi, Tokyo 185-8540, Japan, goto.tomonori.98@rtri.or.jp)

In this study, we propose a new floor structure "suspended floor" as one of the countermeasures against the interior noise of a railway vehicle. The interior noise of a railway vehicle is mainly composed of the transmitted noise and the structure-borne noise from floor panels, where the vibration of the bogie is propagated. Although floor panels are generally supported on joists mounted on the floor structural panels, it is possible to reduce the vibration of floor panels by suspending them from less vibrating points. We investigated the vibrational characteristics of the car body and clarified that the vertical vibration of the side structural panel is less than that of the floor structural panel. Therefore, it was expected that the floor structure in which floor panels are suspended from side structural panels is effective in reducing the vibration of floor panels. The floor structure prototype was fabricated in the test car, and experiments have been made to verify the vibrational reduction effect. As a result, the sound power radiated from the floor panel was reduced by 5-10 dB. In conclusion, the suspended floor structure is effective in reducing structure-borne noise in a railway vehicle.

THURSDAY MORNING, 1 DECEMBER 2016 SOUTH PACIFIC 2, 8:15 A.M. TO 11:45 A.M.

Session 4aPA

Physical Acoustics: General Topics in Physical Acoustics

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Kevin M. Lee, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Contributed Papers

8:15

4aPA1. Stick and slip mechanisms in the acoustic emissions of ice fracture. Rintaro Hayashi, Melisa Yashinski, and John S. Allen (Mech. Eng., Univ. of Hawaii-Manoa, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu)

Arctic ice noise has been qualitatively described in terms of crashing and cracking noises. Ships navigating through icy waters and colliding with icebergs have reported distinctive cracking sounds characteristic to the specific environmental conditions and the location. Many previous field studies of ice acoustics have been done with measurements from isolated, single hydrophones. These have provided useful physical insights into the ambient noise characteristics of ice, but the underlying mechanisms of the sounds are still not well understood. The mechanical properties of ice have been extensively studied in laboratory settings, but few studies have examined the acoustics emissions during fracture with respect these properties. The salinity content of sea ice can significantly influence fracture. The acoustics of ice fracture in water and air is investigated in laboratory using simultaneous acoustic and high speed optical measurements. Also, the mechanical properties of ice samples as a function of salinity were studied with three point bending. A stick and slip mechanism found in fracture of low salinity ice is supported by an Empirical Mode Decomposition of the acoustic signals.

8:30

4aPA2. Determining average flow speed in a scale model smokestack by measuring correlated noise through a long-wavelength acoustic flow meter. JohnPaul R. Abbott (National Ctr. for Physical Acoust., Univ. of MS, 100 Bureau Dr. Stop 8361, Gaithersburg, MD, johnpaul.abbott@gmail.com), Keith A. Gillis, Michael R. Moldover (National Inst. of Standards and Technol., Gaithersburg, MD), and Lee J. Gorny (None, Mountain View, CA)

As part of its Greenhouse Gas and Climate Science Measurements Program, the National Institute of Standards and Technology (NIST) has successfully developed long-wavelength acoustic flowmeters (LWAFs) to measure the average flow speed, V, and the speed of sound, c, for a fluid passing through 1:100 scale test model of a coal-burning power plant smokestack with standard measurement uncertainties of less than 1% and 0.1%, respectively. This improves upon the estimated 5-20% standard measurement uncertainty for existing technologies. With this success NIST has constructed a 1:10 scale test model for further investigation. Current LWAF measurement techniques are reliant on a good signal-to-noise ratios (SNR), which may be problematic for the 1:10 scale test model since the amount of turbulence and difficulty generating high-amplitude low-frequency excitation signals at larger scales increases. Anticipating poorer SNRs for larger scale test models, NIST is investigating the use of the LWAF to correlate low-frequency acoustic noise, flow noise, and other excitation signals as an alternative technique to measure V and c with the goal of obtaining the same standard measurement uncertainties of less than 1% and 0.1%, respectively. The latest results of this ongoing investigation are presented.

8:45

4aPA3. Cut-off frequency criterion for wind noise contributions for wind noise reduction enclosures. JohnPaul R. Abbott (National Ctr. for Physical Acoust., Univ. of MS, 100 Bureau Dr. Stop 8361, Gaithersburg, MD 20899, johnpaul.abbott@gmail.com), Richard Raspet (Dept. of Phys. and Astronomy, National Ctr. for Physical Acoust., Univ. of MS, Oxford, MS), Sandra Collier, W. C. K. Alberts, and John Noble (U.S. Army Res. Labs., Adelphi, MD)

A cut-off frequency criterion for contributions for the wind noise generated from the undisturbed and surface regions for a wind noise reduction device, such as wind fence enclosures and semi-porous fabric domes, is given. The calculations of the measured wind noise inside an enclosure assume that contributions from the undisturbed region and the surface region dominate and are physically restricted to turbulence wavelengths that are larger than the enclosure and the same size or smaller than the enclosure, respectively. The cut-off frequency criterion defines the wavelength where the dominant source of wind noise transitions from interactions in the undisturbed region to the surface region. The transition point is determined by considering when a volume enclosed by a wavelength, V_λ, is equal to 100 times the volume enclosed by the reduction device, V_E. Wind noise calculations following this criterion are presented for a set of semi-porous fabric domes that are 4 m wide and 2.1 m tall, and three cylindrical wind fences that are 5 m wide and 2.9 m tall, 5 m wide and 5.8 m tall, and 10 m wide and 2.9 m tall.

9:00

4aPA4. Numerical determination of parameter sensitivity of transmission loss in atmospheric sound propagation above a general ground surface. Seth D. Hubbard, David Lechner, Joseph F. Vignola (Dept. of Mech. Eng., Catholic Univ. of America, 620 Michigan Ave., Washington, DC 20064, 22hubbard@cua.edu), Teresa J. Ryan, Melissa Hall (Dept. of Eng., East Carolina Univ., Greenville, NC), John A. Judge, and Diego Turo (Dept. of Mech. Eng., Catholic Univ. of America, Washington, DC)

A transmission loss model, similar to the sonar equation, is used to study the transmission loss in atmospheric sound propagation over ground. The model takes into account contributing factors from spreading loss, absorption, wind, temperature gradient, and ground impedance. Numerical predictions obtained with the Crank-Nicolson Parabolic Equation implementation are used to quantify the effects of these factors. The transmission loss sensitivity to these parameters will be presented. These sensitivity findings will be compared to a set of field measurements. An acoustic source is placed in an open field and six horizontally aligned microphones are used to record the signal. Microphones were spaced at regular intervals up to approximately 400 m away from the source and 1 meter above the ground. Temperature, atmospheric pressure, and relative humidity measurements were made concurrent with the acoustic recordings.

9:15

4aPA5. Measurement of atmospheric sound propagation and meteorological parameters to support development of an acoustic propagation model. Melissa A. Hall, Teresa Ryan (Dept. of Eng., East Carolina Univ., 248 Slay Hall, Greenville, NC 27858, hallme14@students.ecu.edu), Seth Hubbard, Joseph F. Vignola, John A. Judge, and Diego Turo (Catholic Univ. of America, N.E. Washington, DC)

Acoustic propagation over long distances is affected by many conditions including atmospheric properties, topography, vegetation, source characteristics, and frequency content. This work presents the results of a set of measurements of acoustic propagation of impulsive sound in the low kilohertz frequency range. These measurements are made over flat ground with limited vegetation at propagation ranges of up to 400 m. At multiple locations along the propagation path, the temperature, pressure, and relative humidity, along with altitude and GPS position are recorded to provide high spatial resolution characterization of the air column in the propagation path. A mast-mounted weather station at the acoustic source provides a record of wind speed and direction at ground level. These measurements are intended to inform related efforts developing a model of acoustic propagation that uses a Crank-Nicolson Parabolic Equation implementation. The measured transmission loss will be compared to preliminary results from the numerical model.

9:30

4aPA6. De-Dopplerization of acoustic signatures. Frank S. Mobley (Human Effectiveness Directorate, U.S. Air Force, 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, frank.mobley.1@us.af.mil) and Brian S. Davis (CDO Technologies Inc., WPAFB, OH)

Shifting energy within and between adjacent fractional octave bands removes Doppler shifting in overflight measurements of acoustic signatures without complex resampling techniques and stationary requirements. Analyzing acoustic measurements with Fourier transforms to obtain spectral information requires complex resampling to overcome stationary signal requirements. A simple shifting of band energy obtained from fractional octave digital filters generates a de-Dopplerized spectrum without complex algorithms. A numerical equation is developed based on the fractional octave center frequency to define the amount of energy shifting required. This equation is applied to a numerical simulation and an overflight measurement to remove the Doppler effect in spectral data. The de-Dopplerization through application of energy shifting accurately removes vehicle motion effects from acoustic measurements without complex resampling.

9:45–10:00 Break

10:00

4aPA7. Focus control of a nematic liquid crystal cell using ultrasound vibration. Yuki Shimizu, Daisuke Koyama (Graduate School of Sci. and Eng., Graduate School of Doshisha Univ., Tataramiyakodani1-3, Kyotanabe, Kyoto 610-0321, Japan, duq0360@mail4.doshisha.ac.jp), Satoki Taniguchi (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto, Japan), Akira Emoto (Graduate School of Sci. and Eng., Graduate School of Doshisha Univ., Kyotanabe, Kyoto, Japan), Kentaro Nakamura (Lab. for Future Interdisciplinary Res. of Sci. and Technol., Tokyo Inst. of Technol., Yokohama Midori-ku, Kanagawa, Japan), and Mami Matsukawa (Graduate School of Sci. and Eng., Graduate School of Doshisha Univ., Kyotanabe, Japan)

Nematic liquid crystals used in optical devices such as liquid crystal displays are controlled by the electric fields. Although indium tin oxide (ITO) is generally employed as the transparent electrode material to apply the electric field, it includes the rare metal indium and needs complicated processes such as sputtering deposition method. Our group has been investigating control techniques of molecular orientation of liquid crystals using ultrasound. In this study, we have applied these techniques to optical variable-focus lenses and the optical characteristics were examined. The ultrasonic liquid crystal cell has a simple structure with no transparent electrodes; it consists of has an annular ultrasound PZT transducer (inner diameter: 30 mm; thickness: 2 mm) and a liquid crystal cell in which a nematic liquid crystal layer (thickness: 25 μm) was formed between two glass substrates (thickness: 0.7 mm) and orientational films. The concentric flexural vibration modes could be generated on the cell at the several resonance frequencies, and the liquid crystal molecular orientation was changed by the acoustic radiation force. The transmitted light through the liquid crystal cell was focused, and the focal length could be controlled by the input signal.

10:15

4aPA8. Standing shear waves in nonlinear gel-like media under static shear stress. Timofey Krit, Shamil Asfandiyarov, and Valery Andreev (Dept. of Acoust., Faculty of Phys., Moscow State Univ., Leninskie Gory, Bldg. 1/2, Moscow, Moscow 119991, Russian Federation, timofey@acs366.phys.msu.ru)

Standing shear waves in a resonator in the form of layer of gel-like medium placed between two rigid plates are studied. The bottom plate is fixed to the vibrator and oscillates in the horizontal direction with a preset amplitude. Two rubber threads attached to the upper plate can displace the plate by a specified value in the horizontal direction. The change in the tension of the threads creates an additional static deformation of the elastic layer resulting in the effective shear elasticity increase. The measured static stress-strain dependence of the elastic layer can be described by the cubic parabola. We measured the dependences of the resonance frequency on the static deformation of the layer. For static deformations of the layer less than 0.3 h (h—is the layer thickness), the resonance frequency increases linearly,

that can be explained by a linear growth of the elastic force of rubber threads. In the deformation range of 0.3-1 h, an additional shift of the resonance frequency caused by the nonlinearity of a gel-like medium appears. The method allows the dynamic nonlinear parameter measurement in gel-like media at the low frequency range.

10:30

4aPA9. Acoustic-vortex interaction in reacting wakes near the global hydrodynamic stability limit. Benjamin Emerson and Tim Lieuwen (Aerosp. Eng., Georgia Inst. of Technol., 635 Strong St., Atlanta, GA 30318, bemerson@gatech.edu)

This study experimentally characterizes the effect of acoustics on the vortex dynamics in reacting wakes. This work is motivated by the problem of combustion instabilities, where the natural combustor acoustics excite hydrodynamic instabilities of the flow that, in turn, induce heat release oscillations. Time resolved PIV, Mie scattering, and chemiluminescence imaging measurements are obtained in a reacting wake facility with acoustic excitation. Measurements and analysis are performed where the acoustic frequency is varied relative to the global mode frequency. We observe that the acoustic waves excite a varicose mode of vortex shedding, which is locked into the acoustic frequency. When the acoustic frequency is close to the natural global hydrodynamic frequency of the wake, we observe resonance, which manifests as very rapid amplification of these shed vortices as they convect downstream. Interestingly, in the resonant case, the vortices stagger from the varicose structure deposited by the acoustic waves and assume a sinuous structure that resembles the hydrodynamic global mode. A linear, local stability analysis, together with a nonlinear analysis, help elucidate the physics that govern the vortex staggering. The study counterintuitively concludes that the resonance, due to the change from varicose to sinuous structure, may help alleviate combustion instability.

10:45

4aPA10. A second-order, perfectly matched layer formulation to model 3D transient wave propagation in anisotropic elastic media. Hisham Assi and Richard S. Cobbold (Inst. of BioMater. & Biomedical Eng., Univ. of Toronto, IBBME, 164 College St., Toronto, ON M5S 3G9, Canada, hisham.assi@utoronto.ca)

Perfectly matched layers (PML) are a well-developed method for simulating wave propagation in unbounded media enabling the use of a reduced computational domain without having to worry about spurious boundary reflections. Using this approach, a compact three-dimensional (3D) formulation is proposed for time-domain modeling of elastic wave propagation in an unbounded general anisotropic medium. The formulation is based on a second-order approach that has the advantages of well-posedness, physical relationship to the underlying equations, and amenability to be implemented in common numerical schemes. However, many auxiliary variables are usually need to described second-order PML formulations. The problem becomes more complex for the 3D case modeling which would explain the dearth of compact second-order formulations in 3D. Using finite element method (FEM), 3D numerical results are presented to demonstrate the applicability of our formulation, including a highly anisotropic medium example. Extension of the PML formulation to model the case of a Kelvin-Voigt viscoelastic medium is also provided. In addition, we will present the extension of this formulation to model the transient wave propagation in an unbounded fluid-solid 3D medium in a similar way to that presented by the authors (JASA, 2016) for the 2D case.

11:00

4aPA11. Performance analysis of finite-difference time-domain schemes for acoustic simulation implemented on multi-core and many-core processor architectures. Ryo Imai, Yukihisa Suzuki (Graduate School of Sci. and Eng., Tokyo Metropolitan Univ., 1-1 Minami-Osawa, Hachioji, Tokyo 192-0397, Japan, imai-ryou@ed.tmu.ac.jp), Kan Okubo, Yuta Katori (Graduate School of System Design, Tokyo Metropolitan Univ., Hino, Japan), and Naoki Kawada (ELSA Japan Inc., Hino, Japan)

There have been many discussions about hardware-accelerations for the large-scale finite-difference time-domain (FDTD) method to deal with various wave dynamics in scientific and engineering computations. Recently, there is growing interest in many-core based parallel computing with graphics processing unit (GPU) or many integrated Core (MIC) accelerators. Therefore, it is worth applying those to large-scale and long-time acoustic simulation for FDTD methods. In this paper, performance analyses are performed for three types of acoustic FDTD schemes, which are FDTD(2,2), FDTD(2,4) and wave equation FDTD (WE-FDTD)(2,2), implemented on GPU, Intel MIC, and multi-core central processing unit (CPU). Here, FDTD(T, S) denotes Tth-order accuracy for the time derivative and Sth-order accuracy for the spatial derivative, respectively. The parallelized FDTD schemes are implemented with Compute Unified Device Architecture (CUDA) and OpenACC, respectively, on GPU, and OpenMP for MIC and multi-core CPU. As a method for performance analysis, we employ the modified roofline model which adds the concept of cache hit ratio to operational intensity. The analysis provides performance comparison results of attainable performance and measured performance for each FDTD scheme and each processor. We performed software optimizations for MIC based on the performance analysis and achieved 78%-91% of attainable performance on MIC.

11:15

4aPA12. Extended optical theorem in elastic solids: Applications to Bessel beam scattering. J. H. Lopes (Exact Sci., Federal Univ. of Alagoas, Arapiraca, Alagoas, Brazil), J. P. Leao, A. L. Baggio, and Glauber T. Silva (Phys., Federal Univ. of Alagoas, Av. Lourival Melo Mota, sn, Maceio, Alagoas 57035-557, Brazil, glauber@pq.cnpq.br)

We derive the extended version of the optical theorem for the scattering of an elastic wave with arbitrary wavefront by an inclusion of any shape embedded in a solid medium. The inclusion can be made of an elastic, viscoelastic, fluid, and rigid material or be an empty cavity. The extended optical theorem relates the extinction cross-section, which is defined as the time-averaged power extracted from the incident wave per incident intensity, in terms of the coefficients of the partial-wave series representation for the incident and scattered waves. With this formalism, the optical theorem for a longitudinal Bessel beam scattering is obtained in a solid for the first time. Additionally, we established the connection between the optical theorem and radiation pressure exerted on a spherical inclusion by a longitudinal and shear plane wave. In conclusion, the developed framework might be useful for beamforming in ultrasound nondestructive test and ultrasound imaging of biological tissue.

11:30

4aPA13. Chaos in a stadium-shaped acoustic cavity. Roger Yu (Phys., New York Inst. of Technol., 1885, New York, NY 10023, ryu@nyit.edu)

A numerical scheme has been developed to solve wave equations for chaotic systems such as stadium-shaped cavity. The same numerical method can also be used for finding wave properties of rectangle cavities with randomly placed obstacles. About 30,000 eigenvalues have been obtained accurately on a normal circumstance. For comparison, we also initiated an experimental study which determines both eigenfrequencies and eigenfunctions of a stadium-shaped cavity using pulse and normal mode analyzing techniques. The acoustic cavity was made adjustable so that the transition from nonchaotic (circle) to chaotic (stadium) waves can be investigated.

Session 4aPPa

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Potpourri (Poster Session)

Frederic Apoux, Chair
Otolaryngology - Head & Neck Surgery, The Ohio State University, 915 Olentangy River Road, Columbus, OH 43212

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

Contributed Papers

4aPPa1. Behavioral and electrophysiological evidence for backward masking. Robert Sears, Silas Smith, Brittany Galvin, and Al Yonovitz (Dept. of Communicative Sci. and Disord., Univ. of Montana, Missoula, MT 59812, al.yonovitz@umontana.edu)

Backward Masking (BM) functions have been shown to relate to age, lead toxicity, and are differentiated in children with language disorders. These functions may be indicative of auditory processing deficits. This study compared BM functions obtained from Evoked Potentials (EP) to behaviorally obtained functions. A stimulus followed by an ISI and a noise masker were presented. All were studied individually in the appropriate temporal alignment. The design of this study allowed observation of the early, middle, and late auditory evoked potentials. For the behavioral task, stimuli were presented with a method of adjustment. This study randomly presented four different stimulus conditions: 1) stimulus alone, 2) noise alone, 3) stimulus and noise, and 4) silence. With a long inter-trial interval (1 s) and high sample rate (25,600 Hz) EP's were obtained for 4000 trials. The stimuli were pure-tones (1000 Hz, 10 ms duration with a Blackman function and brief 100 microsecond clicks of varying intensity and varied ISI. Older and younger subjects were tested. Results indicated that EP's could be arithmetically combined to observe the differential electrophysiological responses and were highly correlated with the behaviorally obtained BM functions.

4aPPa2. Backward masking abilities for speech and tones in persons who stutter. Shriya Basu, Robert S. Schlauch, and Jayanthi Sasisekaran (Dept. of Speech Lang. Hearing Sci., Univ. of Minnesota, 115 SHevlin Hall, Minneapolis, MN 55455, basux045@umn.edu)

We examined eight persons who stutter (PWS) and eight carefully matched adult, control participants for their ability to identify vowel-consonant (VC) syllables and to detect tones in a backward masking paradigm. Speech scores (percent correct) were obtained in quiet and in conditions with a fixed-level broadband masker with a delay of 0 ms and 300 ms. Tonal thresholds (1000 Hz) were obtained using an adaptive procedure in quiet and with the same masker delays as in the speech conditions. The results revealed significantly poorer performance for PWS for all of the speech conditions, including the quiet condition. Tonal thresholds in quiet and for the condition with a 300 ms delay were identical for the two groups, but, on average, the trend was for higher masked thresholds for PWS than for the control group in the 0 ms condition. The poorer speech results are consistent with the idea that PWS may have less distinct representations of phonemic categories that are revealed in a VC task that lacks a language context.

4aPPa3. Within- and between-frequency gap duration discrimination as a function of the standard duration. Hyunsoo Cho (Graduate school of Information Sci. and Elec. Eng., Kyushu Univ., 744 Motooka Nishi-ku, Fukuoka, Fukuoka, Japan, choh@cog.inf.kyushu-u.ac.jp), Kazuhito Ito, Nobuyuki Hirose, and Shuji Mori (Faculty of Information Sci. and Elec. Eng., Kyushu Univ., Fukuoka, Japan)

It is well known that individuals exhibit worse performance during between-frequency gap detection than during within-frequency gap detection. A similar performance difference is also found between within- and between-frequency gap duration discrimination involving undetectably short standard durations. This is reasonable because such discrimination could be considered to be a variant of gap detection. However, because of the paucity of existing data, it is unclear whether frequency differences affects gap discrimination tasks involving longer standard durations. In this study, we measured within- and between-frequency discrimination thresholds in the presence of various standard durations from 1 to 100 ms. As a result, it was shown that the within-frequency thresholds were significantly lower than the between-frequency thresholds when a standard duration of 1-ms was employed, whereas there were no significant differences between them when the standard duration was >50 ms. This implies that the within- and between-frequency gap discrimination processes become equivalent as the standard duration increases.

4aPPa4. Why does gap detection performance in cochlear implant users differs between free-field and direct-stimulation? Etienne Gaudrain (Auditory Cognition and PsychoAcoust., CNRS UMR 5292, Lyon Neurosci. Res. Ctr., Université Lyon 1, UMCG, KNO, Huispostcode BB20, PO box 30.001, Groningen 9700 RB, Netherlands, etienne.gaudrain@cnrs.fr), Pranesh Bhargava, and Deniz Başkent (Dept. of Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen, Groningen, Netherlands)

Gap detection (GD) is used to measure temporal resolution in the auditory system. Previous studies have shown that GD performance in cochlear implant (CI) users is similar to, or sometimes better than, that observed in normal hearing (NH) listeners when simple stimuli are used (e.g., tones, noise). In a recent study employing more ecological speech stimuli, we observed that CI listeners only performed comparably to NH listeners when bypassing the clinical processor and stimulating the implant directly from a research interface. When the stimuli were presented in free field, i.e., through the participants' clinical processor, their GD performance was markedly worse than that of NH listeners. To determine the origin of the discrepancy between direct stimulation and clinical processor results on GD, in the present study, we compared the electrodograms delivered by the implant in the direct stimulation condition to those delivered by the clinical processors of the participants. Our results show that certain aspects of the processing in the implant, likely related to automatic gain control, do reduce the depth of the gap, which is consistent with a reduction in gap detectability. We suggest that GD with natural stimuli could be used to better tune the implant.

4aPPa5. Relationship between the frequency asymmetry of across-frequency gap detection and the temporal asymmetry of cochlear responses. Akihide Takamura (Graduate School of Information Sci. and Elec. Eng., Kyushu Univ., 744 Motooka, Nishi-ku, Fukuoka, Fukuoka 819-0395, Japan, takamua@cog.inf.kyushu-u.ac.jp), Kazuhito Ito, and Shuji Mori (Faculty of Information Sci. and Elec. Eng., Kyushu Univ., Fukuoka, Japan)

Gap detection is often used to estimate the temporal resolution of the human auditory system. Gap detection performance becomes worse when the frequency difference between the leading and trailing markers that delimit a silent gap gets larger. In addition, the across-frequency gap detection threshold is not always constant when the presentation order of the two markers is altered, even if the frequency difference remains unchanged. We suspect that these inconsistencies in gap detection performance are partly caused by cochlear delay; i.e., low frequency signals reach the responding areas of the cochlea later than higher frequency signals. To examine whether across-frequency gap detection thresholds increase when the leading marker is higher in frequency than the trailing marker, as would be expected if the above mentioned hypothesis is correct, we conducted across-frequency gap detection tasks by altering the presentation order of the two markers (marker frequency: from 250 to 8000 Hz). The gap detection thresholds appeared to vary irrespective of the presentation order of the two markers, which does not support the cochlear delay hypothesis.

4aPPa6. Influence of noise on sensitivity to ripple shift and spacing in rippled-spectrum signals. Dmitry Nechaev, Olga Milekhina, and Alexander Supin (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, dm.nechaev@yandex.ru)

The influence of noise on hearing thresholds of ripple shift and ripple spacing in rippled-spectrum signals was investigated in humans. The band-limited noises with rippled spectra were used as signals. The signal spectrum band was 1 octave wide, centered at 2 kHz, the ripple spacing was frequency proportional. With the ripple-shift test, the ripple density was 3.5 cycles per octave and ripple-shift threshold was measured. With the ripple-spacing test, ripple-spacing threshold was measured using a ripple reversal paradigm. A two-alternative forced-choice procedure was combined with adaptive stimulation procedure. Signal levels were 50 or 80 dB SPL. Masker levels varied from 30 to 100 dB SPL. Masked thresholds were obtained for five maskers centered at 0 (on-frequency masker), 0.5, 0.75, 1, and 1.25 oct below the signal (low-frequency maskers). Both on- and low-frequency maskers increased thresholds of both ripple shifts and ripple spacing. The effect of the on-frequency masker depended more on masker/signal ratio than on masker or signal SPL. Effects of low-frequency maskers decreased with increasing the frequency distance between the signal and masker, and its effects became more dependent on masker SPL than on masker/signal ratio.

4aPPa7. Effects of frequency separation and fundamental frequency on perception of simultaneity of the tones. Satoshi Okazaki (Adv. Integration Sci., Chiba Univ., 1-33 Yayoi-cho, Inage-ku, Chiba 263-8522, Japan, okazaki@chiba-u.jp) and Makoto Ichikawa (Letters, Chiba Univ., Inage-ku, Chiba, Japan)

Present study investigated the range of perceptual simultaneity, within which listeners perceive two asynchronous tones as simultaneous, as a function of frequency separation and fundamental frequency of the tones. This function was obtained with frequency separations ranging from 0.03 to 5.00 octaves, and fundamental frequencies ranging from 100 to 800 Hz. Listener's task was to judge whether the two asynchronous pure tones were perceptually simultaneous or not. Results showed a V-shaped function for the perceptual simultaneity range against the frequency separation regardless of fundamental frequency. The perceptual simultaneity range steeply decreased and then gradually increased with the increment of frequency separation. The breakpoint of the V-shaped function was appeared at around the critical bandwidth. This breakpoint was best predicted by CB (Critical Bandwidth) scale, rather than by ERB (Equivalent Rectangular Bandwidth) or octave scales. These findings supported the notion (Okazaki & Ichikawa, 2015, *Proc. Acoust. Soc. Jpn.*) that the perception of simultaneity involves two different processes. These two processes would be explained respectively by the mechanics of cochlear basilar membrane motion, and by phase locking of the auditory nerve fibers.

4aPPa8. Threshold shifts when water was injected into the ear canal reveal the difference among air-, bone-, and cartilage conductions. Tadashi Nishimura, Hiroshi Hosoi (Otolaryngology-Head & Neck Surgery, Nara Medical Univ., 840 Shijo-cho, Kashihara, Nara 634-8522, Japan, t-nishim@naramed-u.ac.jp), Ryota Shimokura (Area of Architecture and Production Design Eng., Shimane Univ., Matsue, Japan), and Tadashi Kitahara (Otolaryngology-Head & Neck Surgery, Nara Medical Univ., Kashihara, Japan)

Cartilage conduction (CC) is a new form of sound transmission which is induced by a transducer being placed on the aural cartilage. To clarify the difference between CC and air or bone conduction (AC or BC), the vibrations of the cartilaginous portion were evaluated by the threshold shifts when water was injected into the ear canal. Seven volunteers with normal hearing participated in this experiment. AC, BC, and CC thresholds at 0.5-4 kHz were measured in the 0%-, 40%-, and 80%-water injection conditions. The water injection elevated the AC thresholds by 22.6-53.3 dB, and the threshold shifts for BC were within 14.9 dB. For CC, when the water was filled within the bony portion, the thresholds were elevated to the same degree as AC. When the water was additionally injected to reach the cartilaginous portion, the thresholds at 0.5 and 1 kHz dramatically decreased by 27.4 and 27.5 dB, respectively. In addition, despite blocking AC by the injected water, the CC thresholds in force level were remarkably lower than those for BC. The current findings suggest that CC is not a hybrid of AC and BC. CC generates airborne sound in the canal more efficiently than BC.

4aPPa9. Predicting the characteristics of wideband absorbance in ears with middle-ear pathologies. Joseph Kei (Audiol. Div., School of Health and Rehabilitation Sci., Univ. of Queensland, Brisbane, QLD 4072, Australia, k.kei@uq.edu.au), Venkatesh Aithal, Sreedevi Aithal (Dept. of Audiol., Townsville Hospital and Health Service, Townsville, QLD, Australia), Shane Anderson, and David Wright (ENT Dept., Townsville Hospital and Health Service, Townsville, QLD, Australia)

Wideband absorbance (WBA), a measure of the proportion of energy absorbed by the middle ear, is an emerging technology to detect middle-ear dysfunction. WBA results in display absorbance as a function of frequency. WBA characteristics in adults with normal middle-ear function show a broad peak between 2 and 4 kHz. While changes in WBA results in patients with middle-ear dysfunction have been described previously, the acoustical properties accounting for those changes have not been investigated. This study investigated the wideband acoustical properties of the middle ear using a spring-mass-friction mechanical system modeling technique. When the system undergoes simple harmonic motion with damping, its frequency of vibration is given by $f = 1/2\pi\sqrt{(k/m)}$, where k and m represent stiffness and mass of the system, respectively. In the presence of middle-ear dysfunction, the k to m ratio changes, resulting in reduced WBA amplitude and a change of the frequency of vibration as shown by a shift of the broad peak of WBA. This study demonstrated the applicability of this modelling technique to account for the changes in acoustical properties of the middle ear in the presence of middle ear dysfunction arising from Eustachian tube dysfunction, otosclerosis, ossicular-chain discontinuity, and cholesteatoma.

4aPPa10. Flexibility of ossicular joints reduces damaging impulse peaks. Peter K. Gottlieb, Yona Vaisbuch (Stanford Univ., Stanford, CA), and Sunil Puria (Eaton Peabody Lab., Harvard Med. School, 243 Charles St., Boston, MA 02114, Sunil_Puria@meei.harvard.edu)

The presence of three distinct middle-ear ossicles is a uniquely mammalian trait whose benefits are still debated. One hypothesis is that the flexibility afforded by the two synovial joints connecting the ossicles provides a protective advantage by dispersing potentially damaging impulsive stimuli before they reach sensitive cochlear structures. The 3D velocity of points along the ossicular chain in unaltered cadaveric human temporal bones ($N = 9$), stimulated with acoustic impulses, was measured in the time domain using a Polytec CLV-3D laser Doppler vibrometer. The measurements were then repeated after fusing one or both of the ossicular joints. Sound transmission was characterized by measuring the amplitude, width, and delay of the impulsive velocity profile as it traveled from the eardrum to the cochlea. In most cases, a delay was noted across either or both of the ossicular joints, and the velocity profile at the stapes was lower in amplitude and broader than at the umbo. Typically, fusing the incus-stapes joint had a small effect, while fusing the incus-malleus joint both significantly increased the amplitude and decreased the width of the velocity profile at the stapes relative to the unaltered case. [Work supported by NIH grants R01 DC 05960 and F31 DC 013943A.]

4aPPa11. Can resonance frequency of the outer ear be measured in neonates using wideband tympanometry? Joseph Kei, Joshua Myers (Div. of Audiol., School of Health and Rehabilitation Sci., Univ. of Queensland, Brisbane, QLD 4072, Australia, k.kei@uq.edu.au), Sreedevi Aithal, Venkatesh Aithal, and Alehandrea Manuel (Dept. of Audiol., Townsville Hospital and Health Service, Townsville, QLD, Australia)

This study investigated the feasibility of measuring the resonance frequency (RF) in healthy neonates' ears using wideband tympanometry (WBT). WBT measures admittance and absorbance as a function of frequency and ear canal pressure. Clinically, RF is the frequency at which the susceptance, the vertical component of admittance, equals zero. In this study, RF was successfully measured in 154 ears (114 newborns) out of 297 ears (182 newborns), which passed a battery of tests including automated auditory brainstem response, 1000-Hz tympanometry and distortion product otoacoustic emissions. The success rate of measuring RF in healthy neonates was 51.9%. The normative data revealed a mean RF of 323 Hz (SD = 67 Hz; range = 240-595 Hz; median = 313.5 Hz; 90% range = 246-440 Hz). No significant gender or ear effects were found. The results of the present study are consistent with the resonance frequency of the outer ear (i.e., elastic ear canal wall) in neonates using a sweep frequency impedance technique (Murakoshi *et al*., 2013). In conclusion, while it is feasible to measure RF of the outer ear in neonates using WBT, the success rate is low.

4aPPa12. Relationship between voicing perception and auditory brainstem responses to stop consonants. Shunsuke Tamura, Kazuhito Ito, Nobuyuki Hirose, and Shuji Mori (Kyushu Univ., 744 Motooka, Nishi-ku, Fukuoka, Japan, Rm. 827, 8th Fl., West Zone II Bldg., Fukuoka 819-0395, Japan, tamuras@cog.inf.kyushu-u.ac.jp)

A number of psychophysical studies have suggested that auditory temporal resolution is related to the voicing boundaries of stop consonants. For example, hearing-impaired listeners with poor temporal resolution have difficulty perceiving voiceless stop consonants. However, there is little neurophysiological evidence for the existence of a relationship between voicing boundaries and auditory temporal resolution. The auditory brainstem response (ABR) is suited to investigations of this relationship because it has proved useful in assessments of auditory temporal processing and speech processing. We conducted a speech identification task using synthesized /da/-/ta/ stimuli in which we varied the voice onset time from 3 to 15 ms in 2 ms steps. There were two types of /da/-/ta/ stimuli whose aspiration noise exhibited high or low power. ABR were measured for the some stimuli across voicing boundaries. The aspiration noises with higher power displayed shorter voicing boundaries and longer peak-to-peak latency between ABR onset responses to the leading noise and vowel parts. This finding suggests that the perception of voicing reflects subcortical temporal processing. [Work supported by JSPS KAKENHI Grant No. 25240023.]

4aPPa13. Comparison of 226 Hz and 1000 Hz tympanometry results in infants. Monika-Maria Oster and Lynne A. Werner (Speech and Hearing Sci., Univ. of Washington, 1417 North East 42nd St., Seattle, WA 98105, mmoster@uw.edu)

Tympanometry with a 226 Hz probe tone is a reliable indicator of middle ear disease in adults and older children. However, it is reported to produce a high rate of false-negatives in young infants. Tympanometry with a 1000 Hz probe tone has been reported to better indicate middle ear dysfunction in infants. However, age-graded norms for 1000 Hz tympanometry are currently not available. Furthermore, the results of 226 Hz and 1000 Hz tympanometry have not been compared in large numbers of infants of different ages. Tympanograms with 226 and 1000 Hz probes were analyzed from 1879 7- to 40-week-old infants to determine whether tympanometric compliance changes with age during infancy, whether 226 Hz compliance predicts 1000 Hz compliance and whether tympanometric screening results obtained with these probe frequencies agree. Preliminary results indicate that infants with 226 Hz compliance of 0.1 mmh had low 1000 Hz compliance. For infants with mean (0.3 mmh) or high (0.5 mmh) 226 Hz compliance, 1000 Hz compliance increased with age. Controlling for age, 226 Hz compliance predicted 1000 Hz compliance. For infants younger than 24 weeks, screening results nearly always agreed. For infants older than 24 weeks, 29.3% of infants passed screening at 226 Hz but failed at 1000 Hz, while almost all who passed, passed at both frequencies.

4aPPa14. Identifying conductive conditions in neonates using wideband acoustic immittance. Joshua Myers (Dept. of Audiol., Townsville Hospital and Health Service, 100 Angus Smith Dr., Douglas, QLD 4814, Australia, myers.josh@gmail.com), Joseph Kei, Alicja N. Malicka (School of Health and Rehabilitation Sci., Univ. of Queensland, Brisbane, QLD, Australia), Sreedevi Aithal, Venkatesh Aithal (Dept. of Audiol., Townsville Hospital and Health Service, Townsville, QLD, Australia), Carlie Driscoll, Asaduzzaman Khan (School of Health and Rehabilitation Sci., Univ. of Queensland, Brisbane, QLD, Australia), Alehandrea Manuel (Dept. of Audiol., Townsville Hospital and Health Service, Townsville, QLD, Australia), and Anjali Joseph (School of Health and Rehabilitation Sci., Univ. of Queensland, Townsville, QLD, Australia)

Wideband acoustic immittance (WAI) is an innovative method of middle ear assessment with significant advantages over currently available clinical tests. Previous large-scale studies in neonates have assessed accuracy against evoked otoacoustic emissions but further research is needed using a more stringent gold standard. The aim of this study was to evaluate the test performance of WAI in neonates against a composite reference standard consisting of distortion-product otoacoustic emissions (DPOAEs) and high-frequency tympanometry (HFT). Five hundred and five neonates were recruited from the maternity ward of the Townsville Hospital to participate in the study. DPOAEs and HFT were performed on each neonate to assess outer and middle ear function. Wideband absorbance and complex admittance (magnitude and phase) were measured from 226 to 8000 Hz in each neonate at ambient pressure using a click stimulus. Best separation between groups that passed and failed the reference standard occurred at frequencies from 1500 to 3000 Hz for absorbance, 1000-2000 Hz for admittance magnitude, and 2000-4000 Hz for admittance phase. The WAI response was modelled using multivariate logistic regression and the model validated with bootstrap resampling. The WAI model accurately identified conductive conditions with an area under the receiver operating characteristic curve of 0.89.

4aPPa15. On the influence of sensorineural hearing loss on the pitch strength of bandpass noise. Maria Horbach, Jesko L. Verhey, and Jan Hots (Dept. of Experimental Audiol., Otto von Guericke Univ. Magdeburg, Leipziger Straße 44, 39120 Magdeburg, Germany, Leipziger Str. 44, Magdeburg 39120, Germany, maria.horbach@st.ovgu.de)

In the perception of environmental noises and in the field of speech and music, tonality plays a crucial role. A tonal character can help the listeners to identify the sound source. However, environmental noises with a tonal character are also considered as especially annoying sounds. The psychoacoustic measure "pitch strength" describes the strength of the tonal sensation evoked by the sound on a scale from weak to strong. For normal-hearing listeners, it was shown in the literature that the pitch strength of bandpass noises decreased with increasing bandwidth when the center frequency was fixed and increased with increasing center frequency when the noise bandwidth was fixed. These results are presumably linked to the frequency selectivity of the auditory system. Since cochlear damage leads to a pathological widening of the auditory filters in the inner ear, an altered perception of pitch strength seems possible. In our study, we investigated whether and to which extent pitch strength is influenced by cochlear damage. For this, we measured the pitch strength of bandpass noise for subjects with sensorineural hearing loss via an absolute scaling method and compared the results to own results and results in the literature of normal-hearing subjects.

4aPPa16. Bandwidth discrimination in normal-hearing listeners. Yuan He and Jennifer Lentz (Dept. of Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, heyuan@indiana.edu)

People's ability to discriminate between different noise bandwidths has not been fully studied. Only a few terms such as "less fluctuating" or "less noisy" are used to describe perceptual changes caused by bandwidth differences. This current study was designed to examine the discrimination threshold of bandwidth in band-pass white noises in normal-hearing listeners. In three experiments, we explored whether center frequency region and base bandwidth would affect discrimination performances, whether discriminating a broader bandwidth is easier than a narrower one, and whether bandwidth discrimination relies on additional cues provided by fixed center frequencies. Thresholds were estimated in a three-interval-three-alternative forced choice task using a three-down one-up adaptive procedure and an oddball paradigm. Visual feedback was given to aid learning about oddball stimulus. We varied the intensity level for sounds within and across trials to avoid the use of intensity cues in all experiments. We also varied the center frequency from interval to interval in one of the experiments to investigate the center frequency cues. In general, we found that normal-hearing listeners may not be very good at discriminating noise bandwidths, especially without the assistance of fixed center frequencies. Large individual differences were observed in all three experiments.

4aPPa17. Context effects in pitch discrimination. Dorothée Arzounian and Alain de Cheveigné (Laboratoire des Systèmes Perceptifs, CNRS, UMR 8342, 29 rue d'Ulm, Paris 75005, France, dorothee.arzounian@ens.fr)

The perceived direction of pitch change between a pair of tones depends on their frequencies, but may also depend on the prior history of stimulation or behavioral response. To quantify the contribution of such context effects, we used a recently developed pitch change discrimination paradigm in which subjects judge the direction of change after each tone of a continuous sequence of tones. Responses were determined primarily by the frequency change of the last pair of tones, but could also be significantly influenced by the frequency of the tone preceding the pair, as well as by the latest response. The magnitude and sign of the effects were subject-dependent, some subjects showing no measurable effect, while in others the weight of the previous stimulus in the decision can go up to half of the weight of the ongoing stimulus. Taking into account these contributions allows for a better estimation of the slope of the psychometric function relating response to frequency change, and thus a better estimate of sensory sensitivity.

4aPPa18. Frequency modulation detection for Mandarin congenital amusics and tone agnosics. Mingshuang Li, Chang Liu (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, limingshuang@utexas.edu), Yun Nan, Wei Tang, Wenjing Wang, and Qi Dong (State Key Lab. of Cognit. Neurosci. and Learning & IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., Beijing, China)

Previous studies showed that congenital amusia affected music pitch and lexical tone processing, as well as the pitch contour discrimination. The current study aimed at exploring whether the deficit in musical and/or tone processing also influenced frequency modulation (FM) detection. Thresholds of frequency modulation detection were examined among three Mandarin Chinese-native listeners groups: tone agnosics (amusics with deficits in lexical tone processing), pure amusics (with normal tone processing) and normal controls. Three acoustic parameters were manipulated: duration (250 and 500 ms), carrier frequency (250 and 1000 Hz), and modulation frequency (2, 8, and 32 Hz). Preliminary results showed that thresholds were higher for pure amusics and tone agnosics than for the control group in most conditions. Auditory model related to the musical and tonal processing for these listener groups will be discussed.

4aPPa19. Tone adjustments, or how an added tone makes a noise less pleasant. Jan Hots and Jesko L. Verhey (Dept. of Experimental Audiol., Otto von Guericke Univ. Magdeburg, Leipziger Straße 44, Magdeburg 39120, Germany, jan.hots@med.ovgu.de)

Several technical sounds contain tonal components which usually arise from rotating parts. Typical examples are sounds originating from turbines or electric engines. These sounds are commonly less pleasant than equal-level sounds without tonal components. To consider this effect on noise pollution, several standards include sections dedicated to tonal components. The standards have in common that they estimate the magnitude of the tones relative to the noise (e.g., tone-to-noise ratio, prominence ratio). Some standards include a second step where a few decibles are added to the measured sound levels, referred to as tone adjustment, to account for the reduction in pleasantness if tonal components are present. The aim of the present study was to measure the tone adjustment. First, the individual masked thresholds of three tonal components were determined. Then, the tonal components were added to the noise at three different levels above the individual masked threshold and a noise alone was adjusted in an adaptive matching experiment to elicit the same loudness or preference as the noise containing the tone. The results are compared to the tone adjustments calculated on the basis of the German standard 45681.

4aPPa20. Effects of age and modality on children's perception of musical meter. Jessica Nave-Blodgett, Erin Hannon, and Joel Snyder (Psych., Univ. of Nevada, Las Vegas, 4505 Maryland Parkway #455030, Las Vegas, NV 89154, erin.hannon@unlv.edu)

When listening to music, experienced adults perceive a metrical hierarchy of stronger and weaker events. Children can tap along with the main beat of music. However, it is unclear whether children can perceive more nuanced and hierarchical aspects of musical meter, such as the greater prominence of events on the downbeat of the measure. We asked 5- to 10-year-old children to provide ratings of fit between musical excerpts and auditory or visual metronomes. Metronomes could match 1) both the beat and measure, 2) the beat but not the measure, 3) the measure but not the beat, or 4) neither the beat nor the measure. Children at all ages gave higher fit ratings to all beat-matching auditory metronomes. For visual metronomes, the same pattern was observed among older children, but younger children's ratings did not vary across conditions. At no age did children give higher ratings to metronomes that matched the measure level, suggesting that, unlike adults, children perceive a main beat but not a hierarchy of beats. These results suggest that children's beat perception develops earlier in the auditory modality than in the visual modality, and listeners' sensitivity to metrical hierarchies may emerge after age 10.

4aPPa21. Effect of temporal asynchrony on children's detection performance in a random-frequency, multi-tonal masking task. Mary M. Flaherty (Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68131, mary.flaherty@boystown.org), Emily Buss (Univ. of North Carolina, Chapel Hill, Chapel Hill, NC), and Lori J. Leibold (Boys Town National Res. Hospital, Omaha, NE)

This study examined the degree to which children can use temporal onset/offset asynchronies to improve performance in a random-frequency, multi-tonal masking task. Normal-hearing school-age children (5-10 years) and adults were tested in two experiments. In Experiment 1, masked thresholds were measured in a 3IFC adaptive procedure. A 1000-Hz pure-tone signal was presented with a random-frequency, 4-tone masker played at 60 dB SPL. Onset/offset asynchrony was manipulated across conditions by fixing the signal duration at 200 ms and examining performance for masker durations of 200, 280, and 400 ms. The signal, when present, was temporally centered in the masker, resulting in asynchronies of 0, 40 and 100 ms. In Experiment 2, asynchrony was manipulated by holding masker duration constant and varying signal duration to be 200, 320, or 400 ms. Preliminary results suggest that children can use temporal onset/offset asynchrony to segregate a fixed-frequency signal tone from a tonal masker. Potential developmental differences in the ability to use a variety of temporal asynchronies in the presence of competing sounds will be discussed.

4aPPa22. What ability is measured in "infant psychophysics"? Lance Nizami (Independent Res. Scholar, Wilkie Way, Palo Alto, CA 94306, nizamii2@att.net)

The magazine *American Scientist* recently celebrated Davida Teller's "forced-choice preferential looking," a psychophysical method that yields a detection/discrimination function for human infant vision. The method was soon adopted and modified for auditory detection/discrimination. Eventually, sufficient data accumulated for a retrospective ("Human auditory development," L. Werner *et al.*). In the auditory trials, an assistant holds an infant exposed to auditory stimuli, and a hidden observer records the infant's behavior, from which an analyst constructs a psychometric function or other quantifier of detection/discrimination. But whose detection/discrimination is quantified, exactly? Consider the roles of the laboratory personnel. The analyst (a receptor) interprets the data (stimulus) supplied by the hidden observer, who is an interpreter (receptor) of the behavior (stimulus) of the infant/infant-holder duo, who constitute a probe (receptor) of the acoustical environment. Altogether, the detection/discrimination ability being inferred is that of the entire Laboratory. In particular, infants' apparent capabilities should improve with the observer's ability to interpret infant behavior, and/or with the observer's expectations, both of which may increase with infant age. Substitute methods, e.g., some already-employed non-invasive physiological recordings such as electro-encephalograms, likewise represent the ability of the Laboratory to probe the environment by way of the infant.

4aPPa23. Womb-like sounds: An intervention for apnea of prematurity. Joanna J. Parga (Neonatology, Univ. of California, Los Angeles, Div. of Neonatal-Perinatal Medicine 10833 Le Conte Ave., MDCC B2-375, Los Angeles, CA 90095, jparga@mednet.ucla.edu), Ravi R. Bhatt (Pediatric Pain and Palliative Care, Univ. of California, Los Angeles, Los Angeles, CA), Kalpashri Kesavan, Myung-Shin Sim (Neonatology, Univ. of California, Los Angeles, Los Angeles, CA), Harvey Karp (Pediatrics, Univ. of Southern California, Los Angeles, CA), Ronald M. Harper (Neurobiology, Univ. of California, Los Angeles, Los Angeles, CA), and Lonnie K. Zeltzer (Pediatric Pain and Palliative Care, Univ. of California, Los Angeles, Los Angeles, CA)

We hypothesized that premature infants exposed to womb-like sounds would show decreased apneic, hypoxemic, and bradycardic events. We believed these findings could be explained by the auditory system exerting influence over the autonomic system and potentially enhancing parasympathetic tone in neonates. Twenty premature infants without comorbidities, 32 to 37 weeks corrected, were their own controls. Infants were exposed to four 6-hour blocks of alternating womb-like sounds with intervening silence. Continuous ECG, respiratory, and oxygen saturation data were collected. Cortisol and alpha amylase salivary samples were obtained at study onset and offset during the 24-hour study period. Intermittent hypoxemic episodes significantly decreased from study onset ($p = 0.03$), along with bradycardic episodes ($p = 0.05$). No reductions in apneic events, cortisol levels, alpha amylase levels, or changes in heart rate variability were found. Exposing premature infants to womb-like sounds reduces hypoxemic and bradycardic events, indicating the potential influence of familiar auditory input on breathing and autonomic output. The sound exposure may provide a low-cost, non-invasive intervention for apnea of prematurity.

4aPPa24. Effects of meaningful or meaningless noise on psychological impression for annoyance and selective attention to stimuli during intellectual task. Takahiro Tamesue and Tetsuro Saeki (Yamaguchi Univ., 2-16-1, Tokiwadai, Ube 755-8611, Japan, tamesue@yamaguchi-u.ac.jp)

The presence of noise during the performance of cognitive task, commonly causes a subjective experience of annoyance, which can lead to a decline in performance. This tendency is stronger for meaningful noise such as a conversation than for meaningless noise such as heating, ventilating, and air-conditioning noise. This paper first focus on the degree of meaningfulness of noise, then discuss the psychological impression of annoyance during auditory or visual cognitive tasks under the meaningful noise or meaningless noise. In addition, we considered how the brain responds on transient event-related potentials (ERPs), elicited by external auditory or visual stimuli, can be measured using electroencephalography (EEG). The present experiment was designed to determine the effects of meaningfulness of the noise on selective attention to stimuli under the odd-ball paradigm. To this end, we examined differences in the N100 and P300 ERPs of these components. The principal component analysis (PCA) was adopted to define a set of components. Our results suggested that degree of meaningfulness of noise has a strong influence on not only the psychological impression of annoyance but also the selective attention to stimuli in cognitive tasks.

4aPPa25. Toward an evidence-based taxonomy of everyday sounds. Oliver C. Bones, Trevor J. Cox, and William J. Davies (Computing, Sci. and Eng., Univ. of Salford, 119 Newton Bldg., Acoust. Res. Ctr., Salford, Greater Manchester M5 4WT, United Kingdom, o.c.bones@salford.ac.uk)

An organizing account of everyday sounds could greatly simplify the management of audio data. The job of an audio database manager will typically involve assigning a combination of textual descriptors to audio data, and perhaps allocation to a predefined category. Retrieval is likely achieved by matching the descriptor to keyword search terms or by browsing through categories. While classification of musical instruments using this type of approach is relatively simple by virtue of the fact that a predefined taxonomy can follow a signal-related hierarchy, non-musical sounds do not necessarily follow such a hierarchy. In addition, classification is made more problematic by the ambiguity of words used to describe everyday sounds. Another area in which the issue of establishing a taxonomy of everyday sounds is particularly pertinent is that of soundscape research; research into soundscapes—acoustic environments as they are perceived—is a relatively new and multidisciplinary area, and as such descriptive terms for everyday sounds are currently used inconsistently. Many existing attempts to taxonomize everyday sounds prescribe categories that are not mutually exclusive, in that everyday sounds could exist in a number of categories; moreover, many are not based on empirical evidence. Here, we present a robust method for creating an evidence-based taxonomy of everyday sounds, involving hierarchical clustering from dimensions generated by correspondence analysis of data from a simple card-sorting and naming procedure.

4aPPa26. Auditory and tactile senses for perceiving the hardness of wood board. Yuri Amimoto and Takayuki Arai (Sophia Univ., 7-1 Kioicho, Sophia University, 4-295C, Arai Lab., Chiyoda, Tokyo 102-8554, Japan, amim0t0.u.ri@gmail.com)

The purpose of the present research is to explore the relationship between auditory and tactile senses on hardness perception. To testify this relationship, this research used a comparative judgment on knocking between two wood boards which have different levels of hardness. The participants were asked to knock the pair of wood boards, and compare the hardness in two conditions: knock with or without auditory information. In the condition with auditory information, sounds were heard directly, or through a sound system, which consists of the frequency filters and sound-proof box. The other condition without auditory information masks the knocking with white noise. In short, the participants compared hardness through these conditions, and the correctness of their judgement were used to measure the contribution of auditory and tactile senses toward perceiving hardness. Consequently, there are differences in judgement between with and without the knocking sound. This suggests the possibility of auditory information affects tactile perception.

4aPPa27. Perception of the weight of a falling ball. Kenji Torii and Hiroshi Hasegawa (Graduate School of Eng., Utsunomiya Univ., 7-2-1 Yoto, Utsunomiya-shi, Tochigi-ken 321-8585, Japan, kenji_torii@n.t.rd.honda.co.jp)

Among perceived qualities, perceived weight is connected with the feeling of luxury. Many studies of methods for controlling the perception of weight have been reported; most concern visual and haptic effects, but auditory effects have not been considered. This presentation reports analysis of how the perception of weight changes with changes of visual and auditory stimuli in videos in which a falling ball strikes a flat plate. We made 6 balls and 4 plates of different materials with the same size but different weights. We made 24 sets (6 balls $\times$ 4 plates) and recorded scenes in which each ball struck each plate. Eleven subjects evaluated the perception of the weight of the ball on a scale of 1 to 7 in all 552 pairs of video comparisons (24×23). Scheffé's paired comparison test revealed that not only visual stimulus but also auditory stimulus can affect the perception of weight.

4aPPa28. A study on the summer forest sound acoustic characteristics using psychoacoustics parameters in South Korea. Seonggeon Bae (Div. of Comput. Media Information Eng., Kangnam Univ., 40, Kangnam-ro, Giheong-gu, Youngin-si, Gyeonggi-do, Korea, Youngin 446-702, South Korea, sgbae@kangnam.ac.kr) and Myungjin Bae (School of Electron. Eng., Soongsil Univ., Seoul, South Korea)

In this paper, summer forest sound acoustic characteristics has been analyzed through psychoacoustics Parameters. Forest sound is closely related to Psychological factors of the human which are analyzed variously. Forest sound is applied to the improvement of depression and anxiety of psychology characteristics. Thus, we study efficient method to apply acoustic properties of forest sound using psychoacoustics parameters.

4aPPa29. Auditory bubbles reveal sparse time-frequency cues subserving identification of musical voices and instruments. Vincent Isnard (IRCAM, 1 Pl. Igor-Stravinsky, Paris 75004, France), Clara Suied (IRBA, Brétigny-sur-Orge, France), and Guillaume Lemaitre (IRCAM, Paris 75004, France, GuillaumeJLemaitre@gmail.com)

Classical timbre studies have modeled timbre as the integration of a limited number of auditory dimensions and proposed acoustic correlates to these dimensions to explain sound identification. Here, the goal was to highlight time-frequency patterns subserving identification of musical voices and instruments, without making any assumption about these patterns. We adapted a "random search method" proposed in vision. The method consists of synthesizing sounds by randomly selecting "auditory bubbles" (small time-frequency glimpses) from the original sounds' spectrograms, and then inverting the resulting sparsified representation. For each bubble selection, a decision procedure categorizes the resulting sound as a voice or an instrument. After hundreds of trials, the whole time-frequency space is explored, and adding together the correct answers reveals the relevant time-frequency patterns for each category. We used this method with two decision procedures: human listeners and a decision algorithm using auditory distances based on spectro-temporal excitation patterns (STEPs). The patterns were strikingly similar for the two procedures: they highlighted higher frequencies (i.e., formants) for the voices, whereas instrument identification was based on lower frequencies (particularly during the onset). Altogether these results show that timbre can be analyzed as time-frequency weighted patterns corresponding to the important cues subserving sound identification.

4aPPa30. A research on understanding of piano players' control of performance behavior in brushing up processes of music piece using Near-infrared spectroscopy. Ayako Matsuo (Graduate School of Adv. Sci. and Technol., Tokyo Denki Univ., 5 Asahi-cho Senju Adachi-ku, Tokyo 120-8551, Japan, matsuoaya.dendai@gmail.com), Takeshi Akita, and Naoko Sano (School of Sci. and Technol. for Future Life, Tokyo Denki Univ., Tokyo, Japan)

In this paper, we are studying about the method in which we could catch the relationship between players' perception of sonic field environment and the following control of playing behavior characteristically. Based on the above way of thinking, we made some measuring experiments on some actual performances in brushing up processes, and analyzed and considered about the difference between experimental results from the view point of the perception and behavior control. Already, it has been shown that measuring the change in frontal lobe brain blood flow using Near-infrared spectroscopy (NIRS) Brain-measuring-apparatus and comparative-analyzing the measured data was different by a variety of psychological conditions. In this paper, adopting the above method, we measured subjects' frontal lobe brain blood flow on two actual performance of the early stage "performance of playing at sight" and the final stage "performance of automated levels" in proficiency-process. After some supplementary interviewing to subjects about their measured practices, we analyzed about control of performance behaviors in some sonic environment. The analysed results show that we could measure two each status of "the perception and behavior control" on actual performance of "performance of playing at sight" and "performance of automated levels" in proficiency-process characteristically.

4aPPa31. Investigating the effect of selective attention and the spatial relationships of competing auditory stimuli on the sound-induced flash illusion. Lindsey R. Kishline, Adrian K. C. Lee (Univ. of Washington, Portage Bay Bldg., Rm. 204, Seattle, WA 98195, lrk4@uw.edu), and Ross K. Maddox (Univ. of Rochester, Seattle, Washington)

The sound-induced flash illusion (SIFI) provides a way of testing multisensory integration through perceptual illusion. Studies disagree on the influence of auditory-visual spatial (in)congruence on SIFI. To better assess the possible influence of spatial proximity, we manipulated the spatial congruence of competing auditory stimuli. Study participants were presented with two timbrally distinct concurrent auditory stimuli of one or two beeps and a visual stimulus composed of one or two flashes. One auditory stimulus always matched the number of visual flashes, and the other did not. The auditory-matching stimulus was manipulated to either be spatially congruent or not with the visual flashes, which could occur centrally or to the left or right. Participants were instructed to report the number of flashes they saw. For half of each session participant's attention was not directed and had no knowledge of flash location on the coming trial, while in the other half there was a visual spatial cue prior to the trial to the location of the visual flashes. We compare and contrast the results with and without spatial attention cueing in order to examine the effect of spatial attention on this multisensory illusion.

4aPPa32. How loudness affects everyday sounds recognition? Patrick Susini, Olivier Houix, Lou Seropian, and Guillaume Lemaitre (STMS Ircam-CNRS-UPMC, 1 Pl. Igor Stravinsky, Paris 75004, France, susini@ircam.fr)

It has been recently shown that loudness affects the perceptual representation of sounds. Following this idea, the present experiment examines whether recognition of everyday sounds is sensitive to study-to-test changes in sound pressure level. In addition, the experiment tests the hypothesis that the sound pressure level changes could hinder recognition more strongly when an encoding task focuses participants on sensory properties of the sounds (e.g., loudness) during the study phase. The study phase used three encoding tasks: sensory (participants rated loudness), semantic (participants categorized sounds in three categories), and control tasks. The test phase measured recognition scores and response times for a list of targets (sounds presented in the study phase) mixed with distractors, both presented at two different levels (L1: the level of the phase study; L2: 15 dB higher for half of the sounds, and 15 dB lower for the other half). Results reveal a significant effect of the level change on recognition scores; recognition is more accurate for sounds at L1. As predicted, recognition is weaker with the sensory encoding task, though the effect is not significant. Those results suggest that loudness is encoded together with semantic attributes in the memory representations of sounds.

4aPPa33. Auditory guided adjustment shows robust loudness constancy. Akio Honda (Dept. of Human Sci. and Cultural Studies, Yamanashi-Eiwa College, 888 Yokone-machi, Kofu, Yamanashi 400-8555, Japan, honda@yamanashi-eiwa.ac.jp), Ayumi Yasukouchi, and Yoichi Sugita (Dept. of Psych., Waseda Univ., Tokyo, Japan)

Perceptual constancy refers to a tendency to perceive an object as having a constant shape, size, and brightness despite apparent changes in these stimulus features. Few studies have systematically and critically examined loudness constancy. We use two adjustment methods to examine it in a natural environment: "auditory guided adjustment", by which listeners play a musical instrument as loud as a model player; and "volume adjustment," by which listeners adjust the loudness of the sound produced by a loudspeaker. The experiments were conducted in the Yamanashi Eiwa College gymnasium, where dark noise was 32-35 dB(A). Stimuli were 2-s musical sounds with G4 (393 Hz) pitch produced by the actual musical instrument performance. The distances separating the model performer and the participant were 2 m, 8 m, and 32 m. Sound pressure levels of the stimuli were set as small, medium, and large corresponding to 60, 75, and 86 dB(A). Participants were asked to produce the sound of the same loudness. It was found that auditory guided adjustment showed more robust loudness constancy than volume adjustment. Transfer to learning through action observation and action-specific perception were discussed.

4aPPa34. Influence of simultaneous grouping mechanisms in the loudness judgment of a multi-tone mixture. Michaël Vannier (Lab. STMS Ircam-CNRS-UPMC, 1, Pl. Igor Stravinsky, Paris 75004, France, michael.vannier@ircam.fr), Nicolas Grimault (CNRS UMR 5292 CRNL, Univ Lyon 1, Lyon Cedex 06, France), Nicolas Misdariis, and Patrick Susini (Lab. STMS Ircam-CNRS-UPMC, Paris, France)

Our main concern is to understand how the perceptual organization of an acoustical mixture into distinct auditory objects (known as an ASA mechanism) can affect the judgments of loudness. Three main questions have been raised: Is the loudness of one auditory object influenced by concurrent objects? How the partial loudness of different auditory objects are combined into one global loudness percept? Does global loudness depend on the perceptual organization of the mixture? In the present experiment, the mixtures consisted of frequency-modulated harmonic complexes. The same or different modulation frequencies have been applied to the odd and even harmonics to ensure that participants perceived either one or two auditory objects, respectively. As a first result, when two objects were well segregated (without spectral masking), the loudness of one object in the mixture equaled the loudness of the same object presented alone. It suggests that spectral grouping mechanisms prevail over loudness. Then, the global loudness pattern obtained for two auditory objects as a function of their partial loudness will be drawn up. Global loudness modeling for two auditory objects will be discussed with regards to the predictions given by classical loudness models. [Research supported by the chair "MOUVIE", carried by the UPMC.]

4aPPa35. Effects of spectral gaps on the contribution of specific frequency bands to the loudness of broadband sounds. Walt Jesteadt, Marcin Wróblewski, and Katie Thrailkill (Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68131, walt.jesteadt@boystown.org)

Contributions of individual frequency bands to judgments of total loudness can be assessed by varying the level of each band independently from one presentation to the next and determining the relation between the change in level of each band and the loudness judgment. Recent studies using stimuli consisting of 15 bands of noise, each two critical bands wide, have shown that frequencies near the lower and upper edges of a broadband noise make a greater contribution to overall loudness than frequencies in the center of the spectrum. In the present study, eight listeners with normal hearing compared the loudness of two 15-band stimuli with average levels of 59 dB SPL per band and of stimuli where 1, 3, 5, or 7 bands were missing from the center of the spectrum. Listeners placed greater weight on the bands at the edges of the spectral gap and the effect was larger for greater gap widths. Edge bands received more weight regardless of absolute frequency and the high-frequency edge of the low bands received greater weight than the low-frequency edge of the high bands. These effects are not predicted by loudness models. [Work supported by NIH.]

4aPPa36. Interindividual differences in directional loudness. Sabine Meunier, Sophie Savel, Jacques Chatron, and Guy Rabau (LMA, CNRS, UPR 7051, Aix Marseille Univ., Centrale Marseille, 4, Impasse Nikola TESLA, CS 40006, Marseille Cedex 13 F-13453, France, meunier@lma.cnrs-mrs.fr)

Sensibility of loudness to sound direction was measured in a two-intervals, 2AFC interleaved-adaptive procedure. We measured the level (at the center of the listener head) of a test sound matching the loudness of a reference fixed at 0° and 65 dB SPL (at the center of the listener head). Azimuths of the test sound varied from 0 to 180° by steps of 15°. Elevations were 15°, 30°, 45°, 60°, and 90°. Both test and reference sounds were wide-band noise. The difference between the levels of reference and test sounds at the point of loudness equality defined the Directional Loudness Sensibility (DLS). A group of 24 listeners showed up large differences in DLS. For the majority, the highest DLSs were around -4 dB. For a few of them DLSs were as large as -9 dB. Some listeners did not exhibit any DLS and could even achieve positive DLSs. The sound pressure levels, measured at the entrance of the ear canal, were related to the DLSs to see whether the interindividual differences in DLS were exclusively due to the filtering by the head or if extra-acoustics factors (such as binaural loudness constancy) might be involved in directional loudness sensitivity.

4aPPa37. Role of static and dynamic interaural differences in the localization of spatially segregated sounds. Daisuke Morikawa (Japan Adv. Inst. of Sci. and Technol., 1-1 Asahidai, Nomi, Ishikawa 923-1292, Japan, morikawa@jaist.ac.jp)

We clarify the role of interaural time difference (ITD) and interaural level difference (ILD) in the localization of spatially segregated sounds. Listeners were asked to distinguish between two sources of white noise having various ITDs/ILDs under head-still and head-rotating conditions. For the former, the segregated rate reached 80% for an ITD of 0.3 ms, when the ITD of other sound source corresponded to the other side. The segregated rate also reached 80% when the ILD between the two noise sources corresponded to an angular difference of 30°. Under head-rotating conditions, the segregated rate reached almost 100%, regardless of the ITDs of the two noise sources. This was because when only one white noise source was used, it was perceived as separate high and low sound images. In confirmations with low- and high-pass noises, the lower-frequency sound image was contained for $<$1.7 kHz and the higher-frequency one was contained for $>$1.2 kHz. In addition, the segregated rate was essentially the same for the ILD under head-still conditions when the ILDs of two noise sources corresponded to the separate side. The sound image was integrated into a single one when the ILDs of two noise sources corresponded to the same side.

4aPPa38. Binaural processing in adults with a history of traumatic brain injury. Christina M. Roup and Julie Powell (Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, roup.2@osu.edu)

An auditory processing deficit is a perceptual issue affecting how the central auditory nervous system understands and makes use of auditory information. Auditory processing deficits are typically associated with the pediatric population; however, anecdotal clinical evidence and the recent focus on central auditory effects of traumatic brain injury (TBI) suggests that auditory processing deficits may be prevalent among adults with a history of TBI. Individuals with symptoms of an auditory processing deficit often score within the normal range on standard clinical assessments. The objectiveof the current study, therefore, was to target measures of binaural processing to identify and characterize auditory processing deficits among adults with a history of TBI. Two groups were recruited: adults with a history of TBI and a control group. Binaural processing was measured with: 1) the Revised Speech Perception in Noise test measured in soundfield; 2) dichotic word recognition without and with low-pass filtering measured in the free recall and directed recall conditions; 3) the 500-Hz masking level difference; and 4) the Listening in Spatialized Noise Sentences Test. Results suggest that individuals with a history of TBI have lower than normal binaural performance.

4aPPa39. Influence of sound source arrangement on the precedence effect. Koji Abe (Akita Prefectural Univ., 84-4 Ebinokuchi Tsutiya, Yuri-Honjyo 015-0055, Japan, koji@akita-pu.ac.jp), Shouichi Takane, Masayuki Nishiguchi, and Kanji Watanabe (Akita Prefectural Univ., Yuri-Honjo, Japan)

As such robot audition, problem of estimating the direction of the sound image from the physical sound signal has high necessity. In order to develop an estimation system of the direction of sound image corresponding to the multiple sound sources, it is necessary to know the behavior of the precedence effect in comprehensive sound source arrangement. So, we investigated the influence of the sound source arrangement on the precedence effect experimentally. The minimum angle between the directions of two sound sources is 30 degrees. Experimental method is a three Alternative Forced Choice Task. The three choices are in the direction of two sound sources and its intermediate. In the case of a bilaterally symmetrical arrangement of sound sources, our experimental results were in good agreement with the findings obtained in previous studies. However, our experimental results show that deviation of the localized direction of sound image is dependent on the sound image arrangement. Particularly, in the case of left-right asymmetrical and front-back symmetrical arrangement, shift of a sound image direction with respect to time difference had a characteristic trend.

4aPPa40. Buildup of contextual plasticity in sound localization. Norbert Kopco and Gabriela Andrejkova (Inst. of Comput. Sci., P. J. Safarik Univ. in Kosice, Jesenna 5, Kosice, Slovakia, norbert.kopco@upjs.sk)

Contextual plasticity (CP) is a form of plasticity in sound localization induced by preceding stimulation. CP is observed as shifts in responses to a click stimulus when, on interleaved trials, the target click is preceded by an identical distractor click coming from a known location. This study examines temporal properties of CP by analyzing behavioral data from two previous experiments [Kopčo, Best, Shinn-Cunningham (2005), ARO Abstract #965; Tomoriová, Kopčo and Andoga (2010), ARO Abstract #827]. In the experiments, the distractor type (single click, train of 8 clicks, or noise), consistency (distractor type fixed within block vs. varying from trial to trial), and location (frontal vs. lateral) were manipulated. The results show that contextual plasticity buildup duration depends on the distractor location. Also, the buildup was stronger with 8-click distractor than 1-click distractor. When distractor type varied, the context type on immediately preceding trial influenced performance in the middle of the run, such that trials following an 8-click context exhibited shifts up to 5° larger than the trials following a 1-click context. No such bi-stable percept was observed in other parts of the run. These results show that CP has complex temporal profile on time scales of seconds to minutes. [Work supported by APVV-0452-12.]

4aPPa41. The role of skull resonance on localization. Michael S. Gordon (Psych., William Paterson Univ., 300 Pompton Rd., Wayne, NJ 07470, gordonm10@wpunj.edu) and Jitwipar Suwangbutra (Psych., William Paterson Univ., New Milford, NJ)

The skull is a dense resonant body that may filter sounds affecting many auditory tasks. Its possible influence has been minimally investigated with localization—a paradigm directly addressed in the current study. Unique spectral information was extracted from the skull recordings of 27 participants. The major lower frequency resonances (93.75—656.25 Hz) and higher frequency resonances (984.38—4781.25 Hz) from each skull were used to filter a set of narrowband noises that were enharmonic with the skulls in this sample. The 75 ms narrowband noises were presented concurrently to both ears or manipulated by adding a 2 ms delay in the left or right channel. Listeners were presented with the stimuli over headphones and were asked to determine its location (left, center, or right). Results indicated that listeners were more accurate localizing low-frequency filtered sounds that were enharmonic with their skull than high frequency peaks. In addition, participants were more accurate localizing to the left and had the most difficulty localizing sounds in the center. These results may be used to suggest the influence of the skull's resonance on localization. Specifically, the resonance of the skull may cause listeners to be more sensitive to sounds of enharmonic spectra.

4aPPa42. Using pupillometry to investigate the better ear advantage. Alan Kan (Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, ahkan@waisman.wisc.edu) and Matthew B. Winn (Univ. of Washington, Seattle, WA)

The dichotic digit span (DDS) task involves listening to and repeating back strings of different digits that are presented to the right and left ears. Listeners with normal hearing typically show better recall of the digits that are presented to the right ear (called the ear advantage, or EA), for reasons that are not fully understood. Conventional measures of intelligibility are sometimes not sensitive enough to reveal a right EA, making it difficult to explore potential explanations for the effect (e.g., whether it is driven by attention or anatomical differences, etc.). In this study, we used pupillometry to investigate the EA because it has been shown to be able to detect subtle differences in speech processing (or listening effort) in cases that are not well distinguished by intelligibility scores. Pupillary responses to various conditions of the DDS test were measured in a group of people with no suspected hearing loss to establish baseline data. Results were at times variable within listeners, which opposes a strict anatomical basis for the EA. Interestingly, the responses showed a sustained period of effort after the end of the stimuli, suggesting the EA could involve working memory rather than the perceptual auditory system alone.

4aPPa43. Behavioral and neural sensitivity to changes in large interaural level differences: Implications for sound localization. Andrew D. Brown, Victor Benichoux (Physiol. & Biophys., Univ. of Colorado School of Medicine, 12800 East 19th Ave., RC1-N, Rm. 7401G, M.S. 8307, Aurora, CO, Andrew.D.Brown@ucdenver.edu), Heath G. Jones (U.S. Army Aeromedical Res. Lab., Fort Rucker, AL), and Daniel J. Tollin (Physiol. & Biophys., Univ. of Colorado School of Medicine, Aurora, CO)

Sound localization acuity depends on source azimuth: minimum audible angles (MAAs) are smallest for sources near the frontal midline. The azimuthal dependence of MAAs can be partially accounted for by the decreasing rate of change in the acoustic cues to sound location (interaural time and level differences [ITDs and ILDs]) with increasing azimuth. Previous work has shown that changes in low-frequency MAAs across azimuth are well-accounted for by changes in physical ITD alone. Here we consider whether acoustical constraints are sufficient to account for the azimuthal dependence of MAAs for high-frequency tones, which are localized on the basis of ILDs. Psychophysical data were obtained from human subjects tested in two different headphone ILD tasks (discrimination and lateralization) and compared to previously published measurements of psychophysical MAAs and acoustical ILDs across azimuth. We conclude that acoustic factors alone are not sufficient to account for the azimuthal dependence of high-frequency MAAs: rather, the increase in MAAs for off-midline positions arises from a combination of reduced ILD information and reduced precision of behavioral ILD sensitivity at large reference ILDs. Further, we show that this psychophysical effect is consistent with the responses of ILD-sensitive neurons, which tend to be weakly modulated at large ILDs.

4aPPa44. Effects of a listener's very slow rotation on sound localization accuracy. Sayaka Tsunokake (Res. Inst. of Elec. Commun., Tohoku Univ., 980-8577, Sendai, Miyagi 2-1-1, Katahira, Aoba-ku, Japan, tsuno@ais.riec.tohoku.ac.jp), Akio Honda (Faculty of Human Sci. and Cultural Studies, Yamanashi Eiwa Collage, Kofu, Yamanashi, Japan), Yôiti Suzuki, and Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Miyagi, Japan)

A listener's head movement is effectively used to accurately localize sounds. By contrast, when a sound stimulus is presented during head rotation, sound localization acuity decreases. Previous studies have shown that sound localization accuracy is degraded even during slow head rotation at 50/s. In this study, we investigated the sound localization accuracy during very slow head rotation from 0.6250/s to 50/s. We measured the detection thresholds (DTs) at the listener's subjective front. The experiment consisted of static and rotation conditions. Listeners were asked to report whether a 30 ms noise burst was presented from the left or right of the subjective front (2 Alternative Forced Choice Task). In the results, the DTs in the rotation condition were larger than that in the static condition. Moreover, DTs seem almost independent of the rotation speed. This suggests that the sound localization resolution at the subjective front is degraded by a listener's passive rotation irrespective of the rotation speed. This is very interesting because a listener only rotates 0.150 during stimulus presentation, which is much less than the DTs in the static condition. Therefore, we investigated the origin of this phenomenon by changing the experimental parameters, including stimulus duration.

4aPPa45. Auditory training and subsequent generalization with speech and non-speech stimuli. Nathaniel Spencer (Wright-Patterson Air Force Base, 711th Human Performance Wing, Wright-Patterson AFB, OH, spencernj80@gmail.com), Eric R. Thompson, Matthew G. Wisniewski, Brian D. Simpson, and Nandini Iyer (Wright-Patterson Air Force Base, Dayton, OH)

Auditory training studies have varied in degree-of-similarity between the training and testing conditions. Some focused testing on the trained task (e.g., Sweetow and Henderson Sabes, 2007), while others addressed whether the training effects generalize to other tasks. The current study used speech and non-speech stimuli to explore how much training in a specific analytical listening task would generalize to performance improvements for another. Naïve listeners participated. Percent correct was measured as a function of signal-to-noise ratio (SNR) in pre- and post-tests with the "multiple-bursts

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same" (MBS) paradigm (Kidd et al., 1994) and the speech-target/speech-masker Coordinate Response Measure (CRM) paradigm. There were three training groups in a between-groups design: 1) training in a masked speech identification task (the CRM) with the competitors varying in f_0 separation (Darwin et al., 2003); 2) training in the MBS task with the protected-region (frequency region with no maskers) bandwidth around the target varying; and 3) no training. Preliminary data describe individual-subject variability in pre-test performance and indicate performance improvements with auditory training. The degree of generalization will be explored for the trained sub-groups. 1) Int J. Audiol. **46**. 374. 2) J. Acoust. Soc. Am. **114**, 2913. 3) J. Acoust. Soc. Am. **95**, 962.

4aPPa46. Talker identification: Effects of masking, hearing loss, and age. Virginia Best (Dept. of Speech, Lang. & Hearing Sci., Boston Univ., Boston, MA 02215, ginbest@bu.edu), Jayne B. Ahlstrom (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC), Christine R. Mason, Elin Roverud, Tyler K. Perrachione, Gerald Kidd (Dept. of Speech, Lang. & Hearing Sci., Boston Univ., Boston, MA), and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

The ability to identify voices was examined under both quiet and masked conditions. Subjects were grouped by hearing status (normal hearing/sensorineural hearing loss) and age (younger/older adults). Listeners first learned to identify the voices of four same-sex "target" talkers in quiet, with a fixed amount of training. On each trial subjects identified which of the four target talkers was asking a brief question (e.g., "What is 2 plus 3?"), and response feedback was given. At test, subjects identified voices (1) in quiet, (2) masked by speech-shaped noise, and (3) masked by a single, unfamiliar same-sex talker. Masked conditions included a range of target-to-masker ratios. The psychometric functions obtained in the speech masker were shallower than those in the noise masker, analogous to typical findings for the task of sentence identification/intelligibility. Both younger and older adults with hearing loss, as well as older adults with normal hearing, generally performed more poorly than younger adults with normal hearing, although large individual differences were observed in all conditions. These preliminary findings suggest that both hearing loss and age may affect the recall of who spoke in conversations, which may contribute to listeners' difficulty in solving the "cocktail party problem." [Work supported by NIH/NIDCD.]

4aPPa47. On the color of speech: The effect of speaking styles on speech recognition by cochlear-implant users. Terrin N. Tamati (Dept. of Otorhinolaryngology/Head and Neck Surgery, Univ. Medical Ctr. Groningen, P.O. Box 30.001, Groningen 9700RB, Netherlands), Esther Janse (Radboud Univ. Nijmegen, Ctr. for Lang. Studies, Nijmegen, Netherlands), and Deniz Başkent (Dept. of Otorhinolaryngology/Head and Neck Surgery, Univ. Medical Ctr. Groningen, Groningen, Netherlands, d.baskent@UMCG.NL)

Speaking styles are a common source of real-life variability that may affect cochlear implant (CI) speech communication. Speech transmitted via a CI is heavily degraded in acoustic-phonetic cues. Therefore, the extent to which (and for what conditions) individual CI users benefit from carefully articulated speech (or are hindered by casually articulated/reduced speech) is unclear. First, we examined the impact of speaking styles on CI speech recognition in favorable listening conditions. CI users and normal-hearing (NH) listeners tested with 8- and 4-channel acoustic CI simulations completed a sentence recognition task with casual and careful sentences (with similar lexical and semantic content) in the quiet. Second, we explored individual performance on this task and its relation with standard speech recognition scores and self-reported real-life hearing abilities. Listeners were overall more accurate on careful sentences than the casual sentences, but casual sentences were disproportionately more difficult than careful sentences for 4-channel stimulations (simulated "poor" CI users) and some CI users. Additionally, CI users with lower self-reported real-life hearing abilities and with lower standard sentence recognition scores were also less accurate on the casual sentences. The results suggest that speaking styles can challenge CI speech recognition, especially for listeners with poor overall recognition skills.

4aPPa48. Minimal local signal-to-noise ratio for glimpsing. Frederic Apoux (Otolaryngol. - Head & Neck Surgery, The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, fred.apoux@gmail.com), Brittney L. Carter, and Eric W. Healy (Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

Glimpsing models assert that speech recognition in noise relies on the ability to detect and combine time-frequency (T-F) units with "usable" speech information. A factor typically used to determine the usability of a unit is the local signal-to-noise ratio (SNR). Determining the SNR below which a unit stops contributing to overall intelligibility, however, is challenging and most approaches have consisted of varying the local SNR criterion. Here we implemented a new approach in which all the units exhibited the same local SNR. In order to prevent glimpsing within a T-F unit, their time and frequency span was lower than the subjects' resolution. Stimuli were created by mixing two sentences at a given overall SNR and retaining only those units with the desired local SNR. This procedure was iterated at various overall SNRs with the same two sentences until we were able to reconstruct 50% of the mixture with only T-F units at the desired local SNR. Sentence recognition was measured for mixtures with a variable (VAR) or constant (CON) local SNR. Our results showed a systematic drop in performance in CON of up to 40% points. More importantly, they suggest a limited contribution of T-F units below -3 dB.

4aPPa49. Comparing speaking fundamental frequency in bilingual and L2-accented speech. Ye-Jee Jung, Soo Bin Lim, Goun Lee, and Seok-Chae Rhee (Yonsei Univ., Yonsei-ro, Seoul 50, South Korea, yj71877@gmail.com)

This study investigates how speaking fundamental frequency (SFF) represents linguistic communities (Korean vs. English) between bilingual and monolingual speakers, and whether speech mode influences SFF. Twelve Korean-English sequential bilingual speakers (6 males, mean age = 26, SD = 4) and twelve Korean L2 learners of English (6 males, mean age = 26, SD = 3) participated in this study. In the controlled speech mode, subjects were recoded producing 22 declarative sentences as well as a short fable; in the spontaneous speech mode, subjects were asked to elaborate on well-known tales. Mean F0 and F0 range were measured for each sentence. The findings established a significant cross-linguistic difference in mean F0, with higher F0 in Korean than in English. We also found that the F0 range is influenced by speech mode, with lower mean F0 and greater F0 range in spontaneous mode. Interestingly, the F0 range differences between the two speech modes were greater in Korean than in English productions only for bilingual speakers. Regarding the L2 effect, inconsistently with the previous findings (Zimmer *et al*., 2016) we did not find a compressed F0 range in L2. These findings confirm that pitch profile can characterize language identity and demonstrate the effect of speech mode on bilingualism.

4aPPa50. Effect of age and hearing loss on spectral integration and speech identification. Mini N. Shrivastav (Commun. Sci. and Special Education, Univ. of Georgia, 570 Aderhold Hall, Athens, GA 30602, mnshriv@uga.edu) and David Eddins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

The goals of the project were to determine how advancing age and hearing loss affect the integration of acoustic features across the audio-frequency spectrum. Several experiments were designed to test listener's abilities to either combine or compare intensive and spectral information in non-speech and speech stimuli across frequency channels, and the association between spectral integration and speech recognition. Listeners included were 16 young normal-hearing, 16 older normal-hearing, and 16 older hearing-impaired individuals. The results indicate that age and/or hearing loss impact aspects of feature encoding as well as the ability to *combine* intensive information across channels. Also, age-related but not hearing-loss-related deficits were observed in spectral integration of temporal information with deficits in encoding as well as the ability to *compare* temporal information across channels. Taken together, the results are consistent with the notion that age- and hearing-loss-related deficits in spectral integration have both peripheral and central components. Results also indicate that the ability to finely discriminate intensive and temporal changes across frequency may not be strongly associated with the ability to recognize

meaningful speech, even though speech is also comprised of dynamic changes in intensive and temporal features.

4aPPa51. Speech-on-speech recognition in children and adults: Effect of whispered speech. Margaret K. Miller, Lori J. Leibold (Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68131, margaret.miller@boystown.org), Lauren Calandruccio (Dept. of Psychol. Sci., Case Western Reserve Univ., Cleveland, OH), and Emily Buss (Dept. of Otolaryngology/Head and Neck Surgery, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

The harmonic structure of speech appears to provide adults with a robust auditory grouping cue, improving speech-on-speech recognition performance. This study evaluated the effect of harmonicity on speech-on-speech recognition in normal-hearing children (5-7 yrs) and adults (19-35 yrs). Masked word recognition was evaluated using a 3AFC procedure with a picture-pointing response. Targets were one-syllable words that lent themselves to illustration, and maskers were samples of two-talker speech. All stimuli were recorded by female talkers in two styles: whispered and natural. Thresholds corresponding to 70.7% correct word recognition were estimated adaptively by adjusting SNR while holding the overall level of the signal-plus-masker at 60 dB SPL throughout testing. Performance was evaluated for whispered target words presented in a whispered two-talker masker and for naturally produced target words presented in a naturally produced two-talker masker. Consistent with previous reports, adults required a more advantageous SNR to achieve criterion performance using whispered compared with natural speech. Children performed more poorly than adults in all conditions, and tended to show a smaller effect of speaking style than adults. These results suggest that children may be less adept than adults at using the harmonic structure of speech to segregate simultaneously presented voices.

4aPPa52. Recognition of vocal emotion in noise-vocoded speech by normal hearing and cochlear implant listeners. Zhi Zhu, Ryota Miyauchi (School of Information Sci., Japan Adv. Inst. of Sci. and Technol., 1-1 Asahidai, Nomi, Ishikawa 9231292, Japan, zhuzhi@jaist.ac.jp), Yukiko Araki (School of Humanities, Kanazawa Univ., Kanazawa, Ishikawa, Japan), and Masashi Unoki (School of Information Sci., Japan Adv. Inst. of Sci. and Technol., Nomi, Ishikawa, Japan)

Chatterjee *et al.* (2015) reported that cochlear implant (CI) listeners have difficulty recognizing vocal emotions due to the limited spectral cues provided by CI devices. Researches on vocal emotion perception of CI listeners have been studying ways to simulate responses of CI listeners by using noise-vocoded speech (NVS) as CI simulations with normal-hearing (NH) listeners. However, it is still unclear whether the results of CI simulations with NH listeners are reliable with regards to CI listeners. This study aims to clarify whether CI listeners can perceive vocal emotion the same way as NH listeners with NVS do. Vocal-emotion recognition experiments were carried out by having both NH and CI listeners listen to original emotional speech and its NVS. The results for CI listeners revealed that they recognized sadness and hot anger more easily than joy and cold anger in both original emotional speech and NVS conditions. Moreover, the results for NH listeners with NVS showed the same trend. The results suggested that the vocal-emotion recognition paradigm using NVS can be used to investigate vocal emotion recognition by CI listeners. [Work supported by the Mitsubishi foundation and Grant in Aid for Scientific Research Innovative Areas (No. 16H01669) from MEXT, Japan.]

Session 4aPPb

Psychological and Physiological Acoustics: Modelling, Measurement, and Technology (Poster Session)

Mathias Dietz, Chair
National Centre for Audiology, Western University, 1201 Western Road, London, ON N6G 1H1, Canada

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

Contributed Papers

4aPPb1. Using Markov chain Monte Carlo algorithms to estimate auditory perceptual abilities. Jennifer Lentz (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, jjlentz@indiana.edu) and Noah Silbert (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

The development of methods to rapidly estimate perceptual abilities has always been a focus of sensory systems research, as many methods can be very time consuming and fairly tedious to the observer. One method recently developed in cognitive psychology is the application of Markov Chain Monte Carlo algorithms to measure perception of stimuli with multiple dimensions, abbreviated MCMCP where the P stands for people. Our original interest was to apply these methods to assess and characterize the tinnitus percept. However, validation of these methods for tinnitus assessment is imperative and development of the algorithm to measure perception of acoustic stimuli could have great advantages for psychoacoustic measurement as well. Here, we present data in which we have applied MCMCP to simultaneously measure the ability to perceive the frequency and bandwidth of a variety of noise bands with different center frequencies and bandwidths. Data suggest that the algorithm converges rapidly on the frequency dimension (typically within 10-15 trials) but not as quickly or reliably on the bandwidth dimension. Our data suggest that it is critical to validate methods used for tinnitus assessment as only certain perceptual dimensions may be well-suited for using such algorithms. These algorithms may also have limitations for the measurement of auditory perceptual abilities.

4aPPb2. A framework for auditory model comparability and applicability. Mathias Dietz (National Ctr. for Audiol., Western Univ., 1201 Western Rd., London, ON N6G 1H1, Canada, mdietz@uwo.ca), Piotr Majdak (Institut für Schallforschung, Österreichische Akademie der Wissenschaften, Vienna, Austria), Richard M. Stern (Carnegie Mellon Univ., Pittsburgh, PA), Torsten Marquardt (Ear Inst., Univ. College London, London, United Kingdom), William M. Hartmann (Michigan State Univ., East Lansing, MI), Dan F. Goodman (Dept. of Elec. and Electron. Eng., Imperial College London, London, United Kingdom), and Stephan D. Ewert (Medizinische Physik und Akustik, Universität Oldenburg, Oldenburg, Germany)

Many computational models of the auditory system exist, most of which can predict a variety of psychoacoustical, physiological, or other experimental data. However, it is often challenging to apply existing third party models to own experimental paradigms, even if the model code is available. It will be demonstrated that model applicability is increased by providing a framework where the model acts as artificial observer performing exactly the same task as the subject (e.g., adaptive staircase procedure). A possible separation of the actual auditory processing of the model from the decision making stage will be discussed, which allows for testing the auditory processing of one model in a variety of experimental paradigms. The framework will consist of a citable data repository providing the required data for the models as well as toolboxes implementing both the auditory models and a variety of experimental paradigms. The model framework will be demonstrated with exemplary binaural models applied to the three most common binaural psychoacoustic paradigms: just noticeable difference (e.g., interaural time difference), tone in noise detection (e.g., binaural masking level difference), and absolute judgment (e.g., sound source localization). Further development of the framework will be discussed.

4aPPb3. Predicting masked thresholds based on physiological models of auditory nerve and inferior colliculus population responses. Laurel H. Carney (Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu) and Virginia M. Richards (Dept. of Cognit. Sci., Univ. of California, Irvine, CA)

Acoustic cues for detection of tones in background noise include envelope, fine-structure, and energy. These cues are transformed in the process of neural coding and are represented in fundamentally different ways by different types of neurons. Here, to study the detection of a tone masked by noise population responses were derived from model auditory-nerve (AN) fibers with low or high spontaneous rates (HSR or LSR) and inferior colliculus (IC) neurons [M. S. A. Zilany, I. C. Bruce, and L. H. Carney, J. Acoust. Soc. Am. **135**, 283-286 (2014) and J. Mao, A. Vosoughi, and L. H. Carney, J. Acoust. Soc. Am. **134**, 396-406 (2013)]. Initial analyses focused on average response rates. Percent correct on two-interval forced-choice tasks was estimated based on the distance between responses to each stimulus interval and a template based on noise-alone responses. Although HSR AN fibers had saturated rates, the addition of a tone reduced low-frequency fluctuations of the time-varying rate, a change encoded by modulation-sensitive model IC cells. This HSR-based code was robust to the roving-level paradigm. While the non-saturated LSR AN fibers responded to added tones with an increase in average rate, in the roving-level paradigm these fibers did not provide reliable information to the model IC cell.

4aPPb4. A neural network model for top-down regulation of sensory plasticity. Yu-Xuan Zhang, Dinglan Tang (National Key Lab. of Cognit. Neurosci. and Learning, Beijing Normal Univ., Yingdong Bldg., Beijing 100875, China, zhangyuxuan@bnu.edu.cn), Ying-Zi Xiong, and Cong Yu (Psych. Dept., Peking Univ., Beijing, China)

Auditory cortex displays remarkable plasticity, but how such plasticity relates to human perceptual learning remains unclear. Here, we present a neural network model that attempts to link the two. The model consists of an encoding layer, a decoding/decision layer, and an interneuron that receives both top-down regulation and bottom-up stimulation. The interneuron, once activated, implements a neural learning mechanism such as reorganization of the tonotopic map or lateral inhibition. During active training, top-down regulation sensitizes the interneuron and initiates the learning process. Depending on the relative influence of top-down regulation and bottom-up stimulation, learning ranges from partially to completely stimulus specific, consistent with empirical observations. For stimulus-specific learning, we have recently shown that passive exposure to an untrained

frequency after, but not before active training induces transfer to that frequency. The model displays similar behaviors regardless of the specific neural learning mechanism (remapping or inhibition) implemented. Overall, the simulation results indicate that auditory cortical plasticity is subject to top-down regulation and that the plasticity, once unlocked, can afford perceptual learning with bottom-up stimulation only.

4aPPb5. The position-variable model at 40. Richard M. Stern (Dept. of Elec. and Comput. Eng., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, rms@cs.cmu.edu)

The position-variable model was first described in 1976 as a means to characterize and predict a variety of binaural lateralization, discrimination, and detection phenomena. The model was motivated by a desire for a more complete understanding of the putative mechanisms by which interaural time and intensity differences were combined, as well as the extent to which results in interaural discrimination and binaural detection experiments are mediated by cues based on subjective lateral position. Predictions of the model were based on the patterns of activity of the display of information proposed earlier by Jeffress and Colburn, based on the auditory-nerve response to the experimental stimuli. The model was originally limited to predictions for a limited set of experiments at 500 Hz. While various extensions of the model have been proposed over the years, they frequently were developed based on incomplete or inconsistent model formulations. This paper describes the capabilities of an extended implementation of the model that addresses a wider variety of experimental phenomena over an extended range of frequencies, all with a single set of assumptions. An implementation of the model will be integrated into the framework for evaluating auditory models described by Dietz *et al.* at this meeting.

4aPPb6. Modeling the effect of olivocochlear efferents on the subcortical envelope following response in humans. Christopher J. Smalt (MIT Lincoln Lab., 244 Wood St., Lexington, MA 02420, Christopher.Smalt@ll.mit.edu), Michael G. Heinz, and Elizabeth A. Strickland (Speech, Lang., & Hearing Sci., Purdue Univ., W Lafayette, IN)

The medial olivocochlear reflex (MOCR), a feedback mechanism that controls outer-hair-cell gain, is thought to provide protection and signal enhancement for a listener in background noise. This listening advantage can be attributed to reduced auditory-nerve firing to noise, which results in an increased dynamic range for encoding signals of interest. The computational model presented here updates the modeling work of Smalt *et al.* (2014) by adding dynamic MOCR characteristics to the more recent "humanized" model version (Zilany, *et al.*, 2014). This MOCR implementation effectively reduces outer-hair-cell gain depending on the stimulus frequency, level, and timing. Human Envelope Following Responses (EFRs; Bharadwaj, 2015) were compared to outputs of the Nelson and Carney (2004) Inferior-Colliculus (IC) model combined with the MOCR auditory-nerve model. Adding simultaneous contralateral noise did not significantly affect the model IC modulation-filter energy at the stimulus frequency, while a 500-ms contralateral noise precursor increased the output by ~3 dB consistent with the human data. Further study of the auditory-nerve model with the MOCR may help to understand human susceptibility to noise-induced hearing loss, the impact of damage to low-spontaneous rate auditory-nerve fibers, as well as to develop new algorithms for hearing aids to restore normal speech perception in noise.

4aPPb7. Testing a computational model for detection of "real-world" sounds. Evelyn M. Hoglund, Niall A. Klyn (Speech and Hearing Sci., Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, hoglund.1@osu.edu), Karl Lerud (Psychol. Sci., Univ. of Connecticut, Storrs, CT), Yonghee Oh (Otolaryngology-Head and Neck Surgery, Oregon Health & Sci. Univ., Portland, OR), Edward Large (Psychol. Sci., Univ. of Connecticut, Storrs, CT), and Lawrence L. Feth (Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

Hoglund (*et al., 2010)* reported the ability of listeners to detect a recorded signal masked by nine different ambient sounds. Signals were selected from 410 recordings of two different helicopters; maskers were selected from nine outdoor field recordings from three different locations. Signal and masker levels were randomized from trial to trial in a 2I,2AFC adaptive procedure to produce psychometric functions. Signal-to-masker ratios (SNR) ranged from -22 to -4 dBA for 70% correct detections. The Hoglund *et al.* results were used to tune a network of non-linear oscillators to optimize its performance. To evaluate the model, recordings of three fixed-wing aircraft were obtained for use as signals in the same nine environmental backgrounds. Signal and masker recordings were pre-processed to ensure that overall RMS voltage measures reflected the sound pressure level measured in a flat-plate coupler at the headphone used by the listeners. Model predictions for complete psychometric functions were obtained before experimental results were available for comparison. This poster presents details of the pre-processing of signals and maskers, and of the adaptive, single-interval procedure used to obtain the SNR for P(C) = 66.5%, 75%, and 87.5% by 16 listeners. Good agreement was found between model predictions and listener performance. [Work supported by a SBIR grant from AFOSR.]

4aPPb8. Modelling the effect of pulse-rate on coding of interaural time differences in listeners with cochlear implants. Suyash N. Joshi and Bastian Epp (Hearing Systems, Tech. Univ. of Denmark, Tech. University of Denmark, Bldg. 352, Ørsteds Plads, Kgs. Lyngby 2800, Denmark, suyashn-joshi@gmail.com)

CIs stimulate the auditory nerve fibers (ANF) with a train of amplitude modulated current pulses. Depending on polarity, the pulses can generate spikes at different sites along the ANF. The latency difference between spikes generated at the central and the peripheral axons was found to be up to 200 μs in cats and up to 450 μs in humans. These timing differences could be the reason underlying the poor performance of CI listeners in ITD perception. A model of ANF responses to electrical stimulation (Joshi et al., 2016), which includes two sites of spike generation along the ANF was used to simulate the ANF responses to constant-amplitude and modulated pulse trains for different pulse-rates. The fidelity of the temporal coding was quantified by calculating the phase-locking value. The results show that an increase in pulse-rate leads to higher uncertainty in the site of spike generation, reduction in phase-locking, and increase in variance of its distribution. This may account for impaired ITD thresholds observed at high pulse-rates. The simulated monaural spike trains are then used predict the ITD discrimination thresholds by imposing delays between the two pulse trains and to identify the factors that affect ITD coding in CI listeners.

4aPPb9. Assessment of the middle-ear assumption of the auditory hazard assessment algorithm for humans. Heath G. Jones, Nathaniel T. Greene, Bankole K. Fasanya, Sean A. Hollonbeck, and William A. Ahroon (Auditory Protection and Performance Div., U.S. Army Aeromedical Res. Lab., Bldg. 6901, Fort Rucker, AL 36362, heath.g.jones2.ctr@mail.mil)

The Auditory Hazard Assessment Algorithm for Humans (AHAAH) is an electrical equivalence model of the human ear designed to reproduce sound transmission from the outer into the inner ear in order to predict potential injury from a given sound exposure. This model was generated and validated using a (mostly) feline model, thus several key assumptions may not hold when adapted to the human. The current project aims to test some of these assumptions, such as the effects of middle-ear muscle contraction (MEMC) on sound transmission and whether the MEMC can be elicited by a conditioning stimulus prior to sound exposure (i.e., a "warned" response). In order to quantify the MEMC, we use laser-Doppler vibrometry to measure tympanic membrane motion in response to a reflex-eliciting acoustic impulse. After verifying the MEMC, we will attempt to classically condition the response by pairing the reflex-eliciting acoustic impulse (unconditioned stimulus, UCS) with a light from a LED display (conditioned stimulus, CS). Changes in the magnitude and/or time-course of the MEMC following repeated UCS-CS pairings will be evidence of MEMC conditioning. Validation of these effects will allow updates to Damage-Risk Criteria to better protect the hearing of Soldiers and civilians exposed to high-level impulsive noises.

4aPPb10. Dimensional interactions in the perception of frequency and bandwidth. Noah H. Silbert (Commun. Sci. and Disord., Univ. of Cincinnati, French East 344, 3202 Eden Ave., Cincinnati, OH 45267, silbernh@ucmail.uc.edu) and Jennifer Lentz (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

As part of a project to develop efficient techniques for measuring the characteristics of tinnitus percepts, multiple listeners exhibited difficulties accurately perceiving the bandwidth of stimuli. Frequency perception seems to dominate bandwidth perception and relative bandwidth judgments seem to interact with frequency judgments in complex ways. In the present work, we use general recognition theory (GRT) to directly probe perceptual interactions between center frequency and bandwidth in noise stimuli. GRT is a multidimensional generalization of signal detection theory, and it allows us to simultaneously model within-category and between-category interactions between dimensions, as well as dimension-specific patterns of response bias. The most common stimulus set for GRT experiments consists of the factorial combination of two levels on each of two dimensions. In order to more fully investigate the perception of frequency and bandwidth, we extend this standard approach to encompass multiple factorial sets of stimuli spanning a larger region of frequency-by-bandwidth space. We will present multilevel Bayesian GRT models fit to frequency-by-bandwidth perception data.

4aPPb11. Measurement of tone-burst-evoked oto-acoustic emissions under various band-noise conditions. Shunsuke Kidani (Hokuriku Univ., 1-1 Taiyogaoka, Kanazawa, Ishikawa 920-1180, Japan, s-kidani@hokuriku-u.ac.jp), Masashi Unoki, and Ryota Miyauchi (Japan Adv. Inst. Sci. and Technol., Nomi, Ishikawa, Japan)

Oto-acoustic emission (OAE), which is reflected by the outer hair cells (OHCs) function, varies depending on the presence of noise. Because it is unclear under what noise conditions the OAE varies, this study investigates the amplitude of the OAEs under various noise conditions. Tone-burst-evoked (TBE)-OAEs were measured at probe frequency with/without band noise in which the bandwidth was 4 ERB_N. The noise was presented at either high or low frequency side of the probe to study how the frequency area affects the TBE-OAEs. In addition, the difference between the noise and the probe frequency was varied to investigate how the presence of the noise inside or outside 1/4 ERB_N from the probe frequency affects the TBE-OAEs. The results indicated that the amplitude of TBE-OAEs decreased when noise was presented to the higher side of the probe frequency and the inside of 1/4 ERB_N from the probe frequency. Interestingly, these results suggest that the amplitude of TBE-OAEs was varied when the noise was presented at the frequency area that has many efferent pathways to the OHCs. [Work supported by JSPS KAKENHI (15K13158).]

4aPPb12. Estimation of auditory compression and filter shape of elderly listeners using notched noise masking. Toshie Matsui (Graduate School of Systems Eng., Wakayama Univ., Sakaedani 930, Wakayama, Wakayama 640-8510, Japan, tmatsui@sys.wakayama-u.ac.jp), Toshio Irino, Hazuki Inabe (Faculty of Systems Eng., Wakayama Univ., Wakayama, Japan), Yuri Nishimura (Graduate School of Systems Eng., Wakayama Univ., Wakayama, Japan), and Roy D. Patterson (Dept. of Physiol., Development and Neurosci., Univ. of Cambridge, Cambridge, United Kingdom)

Notched-noise (NN) masking and the compressive GammaChirp (cGC) auditory filter have recently been used to estimate auditory compression and filter shape simultaneously with young normal-hearing (NH) listeners. This paper reports an extension of the paradigm to measure compression and filter shape in elderly listeners with a range of hearing losses. Six elderly listeners (65-75 year old, male) participated in a compact NN experiment with one probe frequency, 2.0 kHz. There were 21 NN conditions chosen to maximize sensitivity to compression in the mid-level range. The listeners' hearing levels (HLs) at 2.0 kHz ranged from 10 to 50 dB. The cGC filter was fitted to the threshold data to derive level-dependent filter shapes of individuals and their cochlear compression functions. The slope of the central portion of the compression function was normal for all of the listeners except the one with the 50 dB HL, whose slope was close to unity (>0.8 dB/dB). The filter bandwidths were marginally wider than those of young NH listeners. Gathering thresholds in the 21 NN conditions took approximately 3 hours per listener. This is acceptable for laboratory studies with the elderly but still somewhat long for clinical applications.

4aPPb13. Experimental examination of acoustic sensing using super-directivity speaker and super-resolution signal processing with pulse compression technique. Yuya Asakura, Kan Okubo, and Norio Tagawa (Graduate School of System Design, Tokyo Metropolitan Univ., 6-6 Asahigaoka, Hino, Tokyo 191-0065, Japan, asakura.wad3@gmail.com)

Acoustic sensing technology in the air is a promising method to acquire shapes and/or position of a target object. Universal and smart technology in Acoustic sensing, however, cannot be sufficiently established because of various acoustic environmental noises, attenuation effect, etc. Normally, pulse transmission using higher frequency cannot obtain high SNR images due to the atmospheric attenuation, whereas the spatial resolution becomes lower by use of signals in lower frequency range. That is, there is the trade-off between frequency and spatial resolution of transmission signal. In recent years, with the development of computer technologies, a single processing method becomes important for the real-time sensing in air. To overcome these problems, therefore, in this study, we propose and examine acoustic sensing in the air using the super-directivity speaker, super-resolution signal processing, and pulse compression technique (PCT). By applying the super-resolution signal processing and called SCM-MUSIC (FM-chirp based Super resolution Correlation Method—MUltiple SIgnal Classification) method, we can improve the SN ratio and spatial resolution in acoustic sensing. Our experimental results suggest that combination of super-directivity speaker and super-resolution signal processing with PCT enables high-resolution acoustic imaging. Additionally, we discuss the feasibility of far-field high resolution imaging by the proposed method.

4aPPb14. Development of cartilage conduction receiver available in noisy condition. Ryota Shimokura (Shimane Univ., Nishikawatsu-cho 1060, Matsue, Shimane 690-8504, Japan, rshimo@riko.shimane-u.ac.jp), Hiroshi Hosoi, and Tadashi Nishimura (Nara Medical Univ., Kashihara, Japan)

Sound information is known to travel to the cochlea via either air or bone conduction. However, a vibration signal, delivered to the aural cartilage via a transducer, can also produce a clearly audible sound. This type of conduction has been termed "cartilage conduction." The aural cartilage forms the outer ear and is distributed around the exterior half of the external auditory canal. In cartilage conduction, the cartilage and transducer play the roles of a diaphragm and voice coil of a loudspeaker, respectively. A benefit of using the cartilage conduction is to be able to amplify the sound in the canal by pushing the transducer on the cartilage more strongly. And when the pushed ear tragus obscures the canal, the sound pressure in the canal reaches a maximum and the environmental noise is blocked at the same time. In our study, participants could catch a speech clearly in this way in different kinds of noise condition (speech noise, road noise, station noise, and cleaner noise in 50, 60, 70, and 80 dB). This benefit can be applied for a smartphone for workers who have to answer the phone no matter what noisy condition they are in.

4aPPb15. Computer-controlled audiometer's application as an earplug fit-testing tool. J. R. W. Stefanson and William A. Ahroon (Auditory Protection and Performance Div., US Army Aeromedical Res. Laboratory/Hearing Ctr. of Excellence, 6901 Farrel Rd., Fort Rucker, AL 36362, earl.w.stefanson.ctr@mail.mil)

Clinical audiometers are usually coupled with a calibrated pair of TDH-39 headphones, supra-aural earphones that rest on a person's ears. Hearing protector testing with these types of earphones is problematic due to a couple of reasons; the earplug may protrude from the ear canal and come in contact with the earphone thereby alerting the individual that the earplug is likely not inserted adequately or the earphone may impend on the position and affect the seal of the earplug and thereby void any measurement data. Deeper supraural ear-cup cushions were 3D printed for the TDH-39 headphones in order to allow for an insert earplug to be fit-tested without interference. The 3D printed set of ear-cups was calibrated to the computer-controlled audiometer and 20 subjects were tested in the unoccluded and occluded conditions. The differences between the two measurements at each of the seven frequencies tested (i.e., 500, 1000, 2000, 3000, 4000, 6000, and 8000Hz) were averaged across subjects, personal attenuation ratings computed. Results are compared with a commercially available hearing protective device field attenuation estimation system (i.e., fit-check or fit-test system).

4aPPb16. Audio quality in everyday listening situations. Michael Oehler, Thomas Wildenburg (Inst. of Sound and Vib. Eng. (ISAVE), Univ. of Appl. Sci. Dusseldorf, Josef-Gockeln-Str. 9, Dusseldorf 40474, Germany, michael.oehler@hs-duesseldorf.de), and Christoph Reuter (Musicological Dept., Univ. of Vienna, Vienna, Austria)

The quality of digital audio which uses a form of lossy data compression is in the focus of several studies. The experimental focus of these studies includes the fundamental difference between objective and subjective measurement methods (Pocta & Beerends, 2015), test standards (Breebaart, 2015), or signal processing (Khaldi *et al.*, 2013; Jung *et al.*, 2016). But there are only few studies concerning the impact of perceived quality differences in everyday listening situations (ELS). Short excerpts of recent popular music recordings were encoded with different bit rates based on codecs that are commonly used by popular music streaming services. While the encoded files were presented via standard Samsung in-ear headphones, the everyday listening situation was simulated by using binaural recordings of a shopping mall soundscape that was simultaneously presented via Stax electrostatic headphones. In a control condition no soundscape was used. The stimuli were rated by 30 participants in a MUSHRA listening test (ITU-R BS.1534). A repeated measures ANOVA showed that there are significant differences for both conditions concerning the different stimuli (ELS: $F(10,20) = 25.237$, $p = .0001$, $\mu^2 = .927$; no ELS: $F(10,20) = 51.073$, $p = .0001$, $\mu^2 = .962$), but the effect sizes are consistently smaller in the condition with simulated everyday listening situation. The difference between both conditions gets smaller, the lower the bit rates are. A follow-up study additionally includes binaural audio that is currently used by radio stations.

4aPPb17. Noise suppression based on loudness management for hearing aid. Takeru Hiramoto (Dept. of Control and Information Systems Eng., National Inst. of Technol., Kumamoto College, Suya 2659-2, Koshi, Kumamoto 8611102, Japan, ae16hiramoto@g.kumamoto-nct.ac.jp), Nobuhiko Hiruma (Hearing aid and audiological equipment Eng., RION Co., Ltd., Kokubunji, Tokyo, Japan), Hidetoshi Nakashima (Dept. of Control and Information Systems Eng., National Inst. of Technol., Kumamoto College, Koshi, Kumamoto, Japan), and Yoh-ichi. Fujisaka (Hearing Aid and Audiological Equipment Eng., RION Co., Ltd., Kokubunji, Tokyo, Japan)

It is well known that speech intelligibility does not improve even if SNR is improved and noise annoyance decreases by noise suppression for hearing aids, since the noise suppression decreases the level of the desired signal as well as the one of the noise. One solution for the problem is to minimize the speech distortion as much as possible even if a large quantity of residual noise exists. In this study, we propose a noise suppression algorithm based on loudness management to improve speech intelligibility for hearing aid wearer. Our algorithm not only minimizes the speech distortion but also maintains the speech loudness. In the algorithm, the speech spectrum is estimated from the observed signal first, and then the loudness of the estimated speech spectrum is calculated. Simultaneously, the loudness of the observed signal is also calculated and the gains in each frequency band are controlled in order to correspond between these two loudness. In our loudness management, even if the bands are almost filled by noise component, it is not suppressed as much as possible not to reduce barely audible speech component. As results, the noise suppressed signal with low speech distortion is obtained and speech intelligibility is improved comparing with commonly used method.

4aPPb18. Real rooms vs. artificial reverberation: An evaluation of actual source audio vs. artificial ambience in the rear height channels of immersive audio systems. Richard L. King (Music Res., McGIll Univ., 555 Sherbrooke West, Montreal, QC H3A 1E3, Canada, richard.king@mcgill.ca), Brett Leonard (School of Communications, Webster Univ., Webster Groves, MO), and Will Howie (Music Res., McGIll Univ., Montreal, QC, Canada)

Many spatial audio researchers and content producers agree that the best source material for height channels in immersive audio is provided by the capture of actual elevated channels in the room. Particularly for music recording, this technique is preferred as opposed to signal processing, providing a more natural and realistic impression of immersion. While previous work has proven this to be the case in the front channels of various 3D playback systems such as 22.2, the content of the rear height channels has not been specifically evaluated in this respect. Multichannel audio recording, specifically 3D recording, can be a cumbersome task as the channel counts expand—and so the question arises—is it really necessary to capture discrete rear height information? This research compares four height channel capture points are compared to two capture points applied to the front height channels in conjunction with artificial reverberation in the rear channels. A two-part study is employed—the first is a simple ABX test to determine discriminability between the real rooms and the artificially generated version. Part two is a preference test, based on several standard acoustic/perceptual descriptors, revealing the subtle differences between real and artificial rear height channel information.

4aPPb19. Low latency dereverberation with considering speech naturalness for hearing aids. Kotoyo Nozaki, Yusuke Ikeda, Yasuhiro Oikawa (Waseda Univ., 59-407-2, 3-4-1, Ohkubo, Shinjuku, Tokyo 169-8555, Japan, kotokoto3726@asagi.waseda.jp), Yoh-ichi Fujisaka, and Masahiro Sunohara (RION Co., Ltd., Tokyo, Japan)

It is well known that long reverberation makes speech unclear especially for hearing impaired. For a long time, a lot of researches on dereverberation have been studied. However, the computational cost of most of those methods are too expensive for hearing aids. Therefore, we have proposed the simplified dereverberation method, which can estimate reverberation energy from observed signal. In addition, it is also necessary implementing the processing with low latency to avoid an annoyance for hearing aid wearer. Processed signal have to be natural as much as possible to reduce listening effort, although there is performance limit because of simplicity. In this study, we make two proposals in our previous method. First, we apply Frequency-Warped Filterbank (FWF) in analysis and synthesis systems in order to make frequency bands as Bark scaled alignment, and to take the tolerable latency all over frequency for hearing. Second, we introduce the gain control in each frequency to maintain naturalness of processed speech, because reverberation energy varies depending on frequency bands in actual reverberant environments. From subjective experiments, there was a tendency that they prefer this proposal to the previous method.

4aPPb20. A study of a voice evacuation guidance system using a precedence effect. Miyu Hashimoto (Elec. Eng., Nihon Univ., 5-6-18, Hino, Kohnan-ku, Yokohama-shi, Kanagawa-ken 234-0051, Japan, csmy16031@g.nihon-u.ac.jp), Ayumu Osumi, and Youichi Ito (Elec. Eng., Nihon Univ., Chiyoda-ku, Tokyo-to, Japan)

We have developed a voice evacuation guidance system using a precedence effect when a fire breaks out in a building in order to guide refugees to safe place efficiently. This system has already applied to an underground shopping center which has a space like a straight path as an aggressive evacuation guidance system. Further, a method for adapting this system to large interior space is examined and the effectiveness of this system have been identified basically. As a goal of this research, it was intended to develop a highly accurate acoustic design method for designing the voice evacuation guidance system. For realizing this, it is necessary to clarify relationship between acoustic conditions and a sound direction recognition in this system and create a database of these relationship. In this report, as one of important factors of developing this system, we performed an experimental measurement in relation to a recognition time of a composed sound by a preceding sound and a secondary sound. It was studied in particular detail for the secondary sound coming from various angles relative to the preceding sound.

4aPPb21. Characterizing real-world auditory scenes using 360° audio-visual capture. Ashley Foots (Army Res. Lab., P.O. Box 531, Rising Sun, MD 21911, ashleynicolefoots@gmail.com), Kelly Dickerson, and Jeremy Gaston (Army Res. Lab., Aberdeen Proving Ground, MD)

Accurate sound identification is a critical aspect of situational awareness. Environmental sound identification is influenced by physical stimulus dimensions and subjective factors, such as pleasantness or familiarity. *Ecological frequency*, the prevalence of sounds' occurrences in the environment, may also play an important role; however, accurately capturing this information for real-world sounds is difficult. Due to limited information about ecological frequency and the difficulty of accurately characterizing acoustic characteristics of real-world sounds studies do not typically account for this information during stimulus selection. The present study evaluates ecological frequency for environmental sounds collected using a RealSpace Camera, a spherical array of 64 microphones and 5 HD cameras. Environmental sound recordings were captured in various urban and rural settings. Samples were recorded for 30 seconds at five to ten minute intervals for one hour (resulting in up to six samples per session). These multichannel, short-interval recordings were augmented by hour-long single channel samples to enable comparison between long and short term spectro-temporal changes associated with the captured scenes. Human raters then evaluated recorded scenes with and without video confirmation to estimate ecological frequency. This effort is the first step in characterizing the real-world sensory environment and its influence on perceptual performance.

4aPPb22. Externalization of sounds with non-individualized head-related transfer functions and without visual information. Thibaud Leclère, Mathieu Lavandier (LGCB, ENTPE - Université de Lyon, rue Maurice Audin, Vaulx-en-Velin, Rhône 69518, France, thibaud.leclere@entpe.fr), Kevin Perreaut (LGCB, ENTPE - Université de Lyon, Lyon, France), and Fabien Perrin (Auditory Cognition and PsychoAcoust. team, Lyon NeuroSci. Res. Ctr., UCBL - CNRS UMR 5292 - Inserm U1028, Lyon, Rhône, France)

A sound produced by a real source is perceived as "externalized," i.e., outside the head, meaning that it is part of the sound environment of the listener. Conversely, when listening to sounds over headphones, they are generally perceived as "internalized," i.e., inside the head. In the framework of a research project dealing with the evaluation of comatose patients using sounds reproduced over headphones, it is aimed to investigate the influence of the realism of auditory stimulation through some degree of externalization. Experiments were conducted with non-patients (normal-hearing, awake) while conserving the constraints related to the clinical evaluation (non-individualized head-related transfer functions and no visual reference). A first experiment investigated sound externalization using four types of signals (pink noise, music, speech, and bottles clinging) convolved with different non-individualized binaural room impulse responses measured at different azimuths in different rooms. Diotic reference sounds were also tested. A second experiment investigated the influence of direct-to-reverberant ratio and interaural differences by varying these parameters artificially. The results indicated that externalization was affected by both binaural information and reverberation, suggesting the implication of a temporal binaural mechanism. The data were used to seek an acoustical correlate for the degree of non-individualized externalization.

4aPPb23. What is the impact of individualizing head-related transfer functions used for sound externalization? Thibaud Leclère, Mathieu Lavandier (LGCB, ENTPE - Université de Lyon, rue Maurice Audin, Vaulx-en-Velin, Rhône 69518, France, thibaud.leclere@entpe.fr), and Fabien Perrin (Auditory Cognition and PsychoAcoust. Team, Lyon NeuroSci. Res. Ctr., UCBL - CNRS UMR 5292 - Inserm U1028, Lyon, France)

Natural listening can be reproduced via headphones by introducing head-related transfer functions (HRTFs) which represent the temporal and spectral modifications occurring when a sound travels from the source to the listener's ears. HRTFs are individualized, they differ from one listener to the other and strongly depend on their morphology. For practical reasons, it might be complicated to measure HRTFs from each listener and, as a result, non-individualized HRTFs (e.g., measured using a manikin) are sometimes preferred. This study investigated the consequences of this non-individualization on perceived sound externalization. First, individualized HRTFs were measured for a set of listeners in different rooms for sources at different azimuths. Non-individualized HRTFs were also measured at the same positions using a manikin. In a first experiment, listeners evaluated the perceived degree of externalization of sounds convolved with their own HRTFs (individualized condition) or with those measured using the manikin (non-individualized condition). In a second experiment, the effect of equalizing the frequency-response of the headphones was investigated by using either the headphones (equalized or not) or (non-equalized) earphones to deliver the convolved signals. The results of this study are used for a preliminary quantification of the differences between individualized and non-individualized sound externalization.

4aPPb24. Auditory evaluation of head-related transfer functions from multiple databases. Xiao-li Zhong (School of Phys. and Optoelectronics, South China Univ. of Technol., Bldg. No. 18, Wu Shan Rd. No. 381, Guangzhou, Guangdong 510641, China, xlzhong@scut.edu.cn)

In order to rationalize the combined utilization of multiple databases of head-related transfer functions (HRTFs), this work investigates the influence of different HRTF measuring conditions on auditory perception. First, the HRTFs of KEMAR artificial head from five databases were selected, and the diffusion-field equalization was applied to the original HRTFs as a pre-processing method to reduce the differences from different measuring conditions. Then, the spectral deviations among the equalized HRTFs were calculated to objectively evaluate the HRTF's differences from different measuring conditions. Finally, a series of listening experiments (including localization and discrimination experiments) were conducted to subjectively evaluate different HRTF measuring conditions on auditory perception. Results indicate that the different measuring conditions has less influence on the localization performance of HRTFs, suggesting the auditory consistency among multiple HRTF databases in terms of localization; however, only part of HRTF databases are consistent in terms of indiscernibility which is a stricter metric. [Work supported by National Natural Science Foundation of China, No.11474103.]

4aPPb25. A high-resolution head-related transfer function dataset and 3D ear model database. Janina Fels and Ramona Bomhardt (Inst. of Tech. Acoust., Medical Acoust. Group, RWTH Aachen Univ., Kopernikusstraße 5, Aachen 52072, Germany, rbo@akustik.rwth-aachen.de)

Binaural reproduction techniques, which use head-related transfer functions (HRTFs), become more and more popular. Human localization performance using binaural synthesis is, however, often poor due to non-individual HRTFs. To investigate the physical relationship between the ear geometry and the directional HRTFs, the ears and the HRTF datasets of 48 subjects are measured. The detailed ear models are obtained from magnetic resonance imaging scans of the subjects. Two rough scans series of the head and a detailed horizontal scan series, which covers both ears, are taken. The 3D model meshes, which are generated from the MRI scans, allow a very precise determination of anthropometric dimensions which can be used for the individualization of the HRTFs. Additionally, the relationship between the anthropometric dimensions and monaural cues of the HRTFs is studied by the boundary element method. These simulations show which parts of the pinna are involved in case of resonances or destructive interferences.

4aPPb26. Head-related transfer function range extrapolation and near-field compensated high-order Ambisonics. Bo-sun Xie (Acoust. Lab., School of Phys. and Optoelectronics, South China Univ. of Technol., Guangzhou 510641, China, phbsxie@scut.edu.cn)

Head-related transfer function range extrapolation (HRTF-RE) predicts HRTFs at various source distances from the known HRTFs at a fixed source distance and discrete directional grid. Near-field-compensated higher order Ambisonics (NFC-HOA) recreates virtual source at various distances by using a set of loudspeakers arranged on a spherical surface with fixed radius. HRTF-RE and NFC-HOA seem to be two different fields and were studied individually in previous works. The present work analyzes the relationship between HRTF-RE and NFC-HOA. Based on the spherical harmonics-Bessel expansion of sound field, it is proved that HRTF-RE and NFC-HOA are mathematically equivalent and therefore share some common characters and limitations. Based on the equivalence, the interchanging some of the methods used by the two fields is discussed. [This work was supported by the Guangzhou Science and Technology Plan Projects under No. 2014Y2-00021.]

4aPPb27. Individualization of head-related transfer functions by the principle component analysis based on anthropometric measurements. Ramona Bomhardt and Janina Fels (Inst. of Tech. Acoust., Medical Acoust. Group, RWTH Aachen Univ., Kopernikusstraße 5, Aachen 52072, Germany, rbo@akustik.rwth-aachen.de)

A binaural reproduction either uses individual or generic head-related functions (HRTFs). The assessment of individual spatial high-resolution HRTF datasets is very time-consuming and requires a special measurement setup. On the other hand, a generic HRTF dataset from a dummy head can lead to localization errors such as front-back confusions and in-head localizations. In the present work, the generation of individualized HRTF datasets using the principle component analysis and multiple regression analysis from a generic HRTF dataset is discussed. The individualized HRTFs reduce the localization errors by taking individual anthropometric dimensions of the head and pinna into account. The method is applied to 48 individual HRTF datasets to determine the principal components and their weights. The anthropometric dimensions are obtained from the 3D magnetic resonance imaging scans of the 48 subjects. The focus of the work is on the number of necessary principle components with respect to the measurement grid and redundancy of anthropometric dimensions.

4aPPb28. Emphasizing interaural level difference corrects the shift of perceived direction of sound for consumer-grade bone-conduction headphones. Mamoru Iwaki and Yoshiki Chigira (Graduate School of Sci. and Technol., Niigata Univ., 8050 Ikarashi-2-nocho, Niigata 950-2181, Japan, iwaki@eng.niigata-u.ac.jp)

Sounds are perceived not only in air-conduction (AC) manner but also in bone-conduction (BC) one. According to our investigation [Autumn Mtg Acoust. Soc. Japan, 3-1-18, pp. 865-866, 2013; IEEE Int'l Conf. GCCE, pp. 194-197, 2014] about the discrepancy of perceived directions of sound through a BC headphone (TEAC Filltune HP-F200) and an AC headphone (SONY MDR-Z900), the perceived direction through a BC headphone inclined toward the center. In this report, we examined a correction of the shift of perceived direction of sound by emphasizing interaural level difference (ILD). Seven students with normal audibility participated in our experiment. It was carried out in a sound proof room. Sound stimuli were pure-tones of 500 Hz, 1000 Hz, 2000 Hz, and 4000 Hz, with 11 different directions for each stimulus. The sound level through AC headphone was 70 dB. That through BC headphone was adjusted to have the same loudness, for each participant. As a result, the perceived direction through BC headphones was improved. Although the BC headphone used in our experiment produced about 20 dB of ILD in the external auditory meatus, it is difficult to explain that in terms of cross hearing and trans-cranial attenuation, but cartilage conduction hearing. Emphasizing ITD had little improvement of perceived direction.

4aPPb29. Effect of characteristics compensation between air-conduction and bone-conduction headphones on sound localization in an anechoic chamber. Satoki Ogiso (Ph.D. Program in Empowerment Informatics, Univ. of Tsukuba, Tennodai 1-1-1, Tsukuba, Ibaraki 305-8573, Japan, ogiso@aclab.esys.tsukuba.ac.jp), Koichi Mizutani, Naoto Wakatsuki, Keiichi Zempo, and Yuka Maeda (Faculty of Eng., Information and Systems, Univ. of Tsukuba, Tsukuba, Japan)

Bone-conduction (BC) headphones are used when users need to hear provided sounds as well as environmental sounds since they do not occlude the environmental sounds. Sound localization characteristics of the BC sounds are usually worse than that of the air-conduction (AC) sounds. If the frequency characteristics of AC sounds can be reproduced with BC headphones, better sound localization with BC headphones can be achieved. To compensate the frequency characteristics difference between these sounds, a method to measure the frequency characteristics difference is proposed. Also, compensated BC sound is examined by localizing sounds. At first, the difference between AC and BC sounds are measured by tuning their amplitudes and phases to cancel at perceiving sound. The measurements were conducted for 30 frequencies in the audible range. Second, localization characteristics are measured in an anechoic chamber. The sounds are presented from AC and BC headphones using the corresponding head related transfer functions in the horizontal plane. Especially for the BC headphone, the characteristics difference between AC and BC are compensated. Based on single-user experiment, characteristic peaks were confirmed in the frequency characteristics difference. The more detailed experimental results with multiple number of users and discussions will also be provided.

4aPPb30. Investigating integration between auditory and visual location information presented in an augmented-reality device. Hiraku Okumura (R&D Div., Yamaha, 203 Matsunokijima, Iwata, Shizuoka 438-0192, Japan, hiraku.okumura@music.yamaha.com), Khiwadkar Sushrut, and Sungyoung Kim (Rochester Inst. of Technol., Rochester, NY)

Effective rendering technique of auditory information for an augmented reality (AR) device has been investigated. Researchers and industries are searching for new applications incorporating AR and Virtual Reality (VR) technologies, which provide enhanced user-experience of interactivity and intimacy. To utilize these benefits best, homogeneous integration between visual and auditory information is important. To date, the binaural technology based on head-related transfer function (HRTF) has been used to create immersive and three-dimensional audio objects for VR and AR devices. In this study, we compared the HRTF method with a stereophonic panning method that only controls Inter-aural Level Difference (ILD) for an AR device (a smart see-through glass) and investigated the precision of auditory information required for a coherent representation with a target visual image at the locations of 0 degree, -5 degrees, and -10 degrees in a counter clockwise. Auditory stimuli were rendered to have target locations in the horizontal plane from +45 degrees to -45 degrees with a 5-degree interval. In the subjective evaluation, each participant answered whether a randomly given auditory location was matched to a target location of the visual image. The results showed that two audio rendering methods did not produce significant difference in creating integrated perception in an AR device.

4aPPb31. Subjective attributes of realistic sound fields: What makes acoustic environments complex? Adam Weisser and Jorg M. Buchholz (Linguist, Macquarie Univ., National Acoust. Labs., Australian Hearing, Level 4, 16 University Ave., NSW 2109, Australia, adam.weisser@nal.gov.au)

Real-life acoustic environments are commonly considered to be complex, because they contain multiple sound sources in different locations, room reverberation, and movement—all at continuously varying levels. In comparison, most laboratory-generated sound fields in hearing research are much simpler. However, the continuum between simple and complex fields is likely to be a function of multiple instrumental as well as perceptual factors. Although most of the applied knowledge has been gathered about simple, controlled, and synthetic acoustic environments, inferring listeners' performance and perception from that to realistic scenarios is non-trivial. In order to clarify some of these relationships, subjective responses to twelve virtualized acoustic environments were examined using exploratory questionnaires with both descriptive and rating questions. These were specifically designed for the purpose of identifying attributes that may correlate with perceived acoustic complexity. Environments were recorded and processed using higher-order ambisonics, and reproduced in a 41-loudspeaker array in an anechoic chamber. Eleven environments spanned from a quiet library to a loud food court and diffuse noise was used as a control environment. Attributes included questions about busyness, variability, reverberance, annoyance, and others. The results enable us to break down the perceived complexity to its constituent factors.

4aPPb32. Simulation of frequency characteristics of bone conduction by own speech. Takayoshi Nakai and Kenta Suzuki (Faculty of Eng., Shizuoka Univ., 3-5-1 Johoku, Naka-ku, Hamamatsu, Shizuoka 432-8561, Japan, nakai.takayoshi@shizuoka.ac.jp)

When a speaker himself/herself hear his/her own speech, it is different from recorded speech. It is known that it is due to bone conduction. But we measured sound at the entrance of the ear, and it is shown that speech heard by speaker himself/herself is almost agreement with sound at the entrance of the ear at more than 1 kHz. This time, we simulate frequency characteristics of bone conduction. The vocal tract have loss factor of air viscosity and heat conduction of the wall, and it is assumed that the ear drum is vibrated by vibration of the vocal tract. These results are reported.

4aPPb33. Control of a sound image to the rearward direction using band-stop filters and laterally located loudspeakers. Tomomi Hasegawa, Satoshi Oode, Kazuho Ono (NHK STRL, 1-10-11, Kinuta, Tokyo, Setagaya-ku 157-8510, Japan, hasegawa.t-eo@nhk.or.jp), and Kazuhiro Iida (Chiba Inst. of Technol. / NHK STRL, Chiba, Japan)

The ability to control the sound image to the rearward direction using front or lateral loudspeakers is an important step to popularize multichannel 3D audio systems in the home environment. Rear "directional bands" were found for narrow-band signals which are perceived only in the rearward direction, but any kind of directional bands for wide-band signals, which are perceived only in the rearward direction, have not been reported. It is known that spectral notches in head-related transfer functions are important cues for a sound image localization in the median plane. In the present study, the possibility of sound image control to the rearward direction was investigated using wide-band noise of which frequency components adjacent to the rear directional bands were eliminated. A total of 56 kinds of band-stop filters were used to eliminate one to the ten consecutive 1/3-octave bands with center frequencies of 1.6 to 12.5 kHz. In-phase stimuli were emitted from a pair of loudspeakers located at azimuth angles of $\pm 90^\circ$ in an anechoic room. The results showed that the probability of perceiving a sound image to be in the rearward direction was 95% when consecutive frequency bands, which were customized to individual subject, were eliminated.

4aPPb34. Psychophysical investigation of binaural spatial-audio music perception with varying degrees of realism. Craig T. Jin (School of Elec. and Information Eng., Univ. of Sydney, Maze Crescent, Bldg. J03, Sydney, NSW 2006, Australia, craig.jin@sydney.edu.au), Nicolas Epain (Inst. of Res. and Technol., B<>Com, Cesson-Sévigné, Rennes, France), and Mengyao Zhu (Commun. and Information Eng., Shanghai Univ., Shanghai, China)

A psychoacoustic listening test is conducted to explore the influence of varying degrees of realism on binaural spatial-audio music perception. The specific factors that are varied in combination are: (1) whether or not head-tracking is used; (2) whether or not individualized Head-Related Impulse Response filters are used; (3) the number of sound-track channels: mono, stereo, and 5.2 surround; and (4) whether or not reverberation is used. Experiments are repeated a number of times with listeners requested to make judgement comparisons regarding: externalization, quality of the frontal image, timbre, and preference. Open headphones (AKG 1000) are used so that listeners can compare the various binaural spatial-audio rendering conditions with standard non-binaural headphone presentation and also loudspeaker presentation. The results indicate the impact of varying degrees of realism on various aspects of binaural spatial-audio music perception.

Session 4aSA

Structural Acoustics and Vibration and Noise: Building Vibration Analysis, Measurement, and Mitigation

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

Hasson M. Tavossi, Cochair
Physics, Astronomy, and Geosciences, Valdosta State University, 1500 N. Patterson St., Spring Valley Cir, Valdosta, GA 31698-0055

Yoshihiro Hirao, Cochair
Noise & Vibration Division, Kobayasi Institute of Physical Research, 3-20-41, Higashi Motomachi, Kokubunji 185-0022, Japan

Chair's Introduction—7:55

Invited Papers

8:00

4aSA1. INCE/J Vibration Measurement Manual for Buildings—Part 1: A commentary on the manual for vibration measurement. Shigenori Yokoshima (Kanagawa Environ. Res. Ctr., 1-3-39, Shinomiya, Hiratsuka, Kanagawa 2540014, Japan, yokoshima@k-erc.pref.kanagawa.jp)

Vibration Regulation Law in Japan has regulated vertical ground vibration caused by road traffic, operation of a factory machine, and construction work since 1976. However, inhabitants perceive vibrations not on the ground but in buildings which are residential spaces. For the purpose of assessing building vibrations, technical committee on Environmental Vibration Evaluation in the Institute of Noise Control Engineering (INCE/J) organized Vibration Manual for Buildings. The manual is desired to provide technical materials contributing to effective measures for the vibrations to be solved. To achieve the aim, the manual concerns the method of measurement of the following vibration characteristics: propagation of ground vibrations, transmission of house vibrations, and vibration exposures to inhabitants according to vibration sources. In addition, the accumulation of detailed measurement data contributing to the revision of vibration index in the future is an important issue. Specifically, items of the manual are the following: scope, measurement (measurement unit, instrument, location direction, time zone, and notices) and calculation of measurements. For example, measurement unit is based on one-third octave band analysis. The vibration shall be measured at the location in building where the inhabitants perceive highest magnitude of the vibration.

8:20

4aSA2. INCE/J Vibration Measurement Manual for Buildings—Part 2: On the assessment based on the vibration. Yasunao Matsumoto (Dept. of Civil and Environ. Eng., Saitama Univ., 255 Shimo-Okubo, Sakura, Saitama 338-8570, Japan, ymatsu@mail.saitama-u.ac.jp)

This paper discusses the assessment of building vibration in terms of human responses based on vibration measurements specified by the Vibration Measurement Manual for Buildings developed by the Technical Subcommittee on Environmental Vibration Evaluation of the Institute of Noise Control Engineering of Japan. As stated in some of the current international and national standards, building vibration could cause adverse comments from occupants when the vibration magnitude is in excess of human perception only slightly. For the assessment of building vibration based on the measurement manual, therefore, a range of human vibration perception threshold was derived from the previous studies that determined the perception thresholds in experiment involving human subjects. The vibration perception threshold range implies a range of vibration magnitude for different frequencies that may correspond to lowest vibration magnitudes people with different sensitivities to vibration can detect. The vibration magnitude within or in excess of the perception range may be assessed as a magnitude that can be perceived by occupant at different possibilities. The assessment with the vibration perception threshold range does not necessarily show if the building vibration to be assessed has a magnitude high enough to cause adverse comment.

8:40

4aSA3. INCE/J Vibration Measurement Manual for Buildings—Part 3: Measurement example of building vibrations. Yoshihiro Hirao (Noise & Vib. Div., Kobayasi Inst. of Physical Res., 3-20-41, Higashi Motomachi, Kokubunji, Tokyo 185-0022, Japan, hirao@kobayasi-riken.or.jp)

The environmental impact on humans in buildings is caused by ground-borne vibration from rail transit systems, road traffic, construction sites, and industrial plants. To estimate the effect of vibration on humans, it is necessary to measure the tri-axial vibrations in the vertical and horizontal directions in buildings specified in ISO 2631-2:2003 or "Vibration Measurement Manual for Buildings" publicized by the Technical Subcommittees on Environmental Vibration Evaluation of The Institute of Noise Control Engineering of Japan. In this measurement example of building vibrations, vibration measurement was carried out for eight hours in a three-story, wooden detached house adjacent to a construction site. The residents perceived discomfort or annoyance from the vibration caused by the demolition work at the construction site. Five vibration measurement devices recorded simultaneously the vibration accelerations on the floor of each story, the concrete substructure and the asphalt pavement surface near the substructure. The 1/3 octave band vibration accelerations of each floor were estimated by the vibration perception threshold range of the vibration measurement manual. The vibration amplifications caused by building structural resonance was evaluated as ratios and level differences in the vibration accelerations measured at the ground near the substructure and floors in the houses.

9:00

4aSA4. INCE/J Vibration Measurement Manual for Buildings—Part 4: Measurement of ground propagation characteristic and long-term measurement in building for road traffic vibration. Ryosuke Kazama, Kodai Yamashita, Hiromitsu Umayahara, and Toshiyuki Hasumi (RION Co., Ltd., 3-20-41 Higashimotomachi, Kokubunji, Tokyo 185-8533, Japan, r-kazama@rion.co.jp)

Vibration Measurement Manual for Buildings, issued by Technical Subcommittees on Environmental Vibration Evaluation of The Institute of Noise Control Engineering of Japan, describes how to measure and evaluate vibration. According to the manual, vibration level and 1/3 octave band vibration acceleration levels, which are calculated from tri-axial acceleration waveforms, are used as measurement values. For the values of each axis, we extracted the 10 maximum values of each band level around the time when vibration levels reached maximum by vibration occurrences. The final measurement result is arithmetic average of the maximum 10 data. Some measuring instruments including vibration level meters, data recorders and waveform analysis software are necessary to obtain the result. To show how to measure vibration and how to operate measuring instruments based on the manual, we report two measurement examples about road traffic vibration which is easily perceptible. First, we measured a ground propagation characteristic at multi points along straight line from vibration source. Second, we conducted a long-term vibration measurement in a building, assuming low occurrence of vibration which caused complaints. As a result, we found that the level of certain frequency bands in vertical direction reached the vibration perception threshold range.

9:20

4aSA5. INCE/J Vibration Measurement Manual for Buildings—Part 5: Mounting method for vibration pick-up on ground-borne vibration measurements. Kozo Yamamoto (Tech. Res. Inst., Toyo Construction, 1033-1, ukeryo, Miho-mura, Inashiki-gun, Ibaraki 300-0424, Japan, yamamoto-kozo@toyo-const.co.jp), Chikara Oishi (Architectural Environ. Res. and Eng. Inc., Yokohama, Kanagawa, Japan), and Yoshihiro Hirao (Noise and Vib. Lab., Kobayasi Inst. of Physical Res., Kokubunji, Tokyo, Japan)

In order to avoid the negative effect of contact resonance caused by the vibration system consisting of the vibration damping spring (ground surface) and the mass (vibration pick-up) in ground-borne vibration measurements, the vibration pick-up is put on the hard ground surface. However, when it must measure on the soft ground, after removing dry soil and grass, it is necessary to fully tread on the ground surface. In addition, the mounting method using the vibration pick-up stand with quad-peg is recommended in "The Vibration Measurement Manual for Buildings" proposed by the Technical Subcommittee of Environmental Vibration Evaluation of the Institute of Noise Control Engineering of Japan. We verified that new single-peg type vibration pick-up stand was a practical and useful tool via comparison with quad-peg type vibration pick-up stand. Moreover, we verified that the suitable hardness treading on the ground surface was acquired from the support strength of ground surface.

9:40

4aSA6. Vertical sensory evaluation on residential floor by traffic vibration. Toru Matsuda (Daiwa House Industry, 6-6-2, Sakyo, Nara, Nara 631-0801, Japan, toru.matsuda@daiwahouse.jp), Katsuo Inoue, and Ryuta Tomita (Nihon Univ., Chiyoda, Tokyo, Japan)

If vibration failures have occurred in building caused by traffic, in Japan, it is often evaluated by "Guidelines for the evaluation of habitability to building vibration." This evaluation method has to be collated acceleration maximum value of 1/3 octave band analysis on the evaluation curve, to evaluate the response acceleration of adjacent frequency bands to reference curve. This performance evaluation curves are presented from experimental results based on sine vibration, under real environment, random vibration containing multiple frequency components. Therefore, it is necessary to consider the influence of the frequency components other than determination frequency, because it is considered to affect the vibration sense. Therefore, in this paper, as a basic study, we examined the good evaluation physical quantity of corresponding to vibration sense from relationship of the vibration sense and vibration response physical quantity for traffic vibration of railway and road. As a result, in the range of more than feeling the vibration, in the case of the narrowband vibration response, which is equal to broadband vibration response, the broadband vibration response has been suggested the increase in perception and discomfort. That is, the frequency characteristics other than the determination frequency have been suggested as one of the factors affecting vibration sense.

10:00–10:15 Break

10:15

4aSA7. Spectral analysis of absorption characteristics of particulate dampers to mitigate building vibration. Hasson M. Tavossi (Phys., Astronomy, and GeoSci., Valdosta State Univ., 1500 N. Patterson St., Spring Valley Cir, Valdosta, GA 31698-0055, htavossi@valdosta.edu)

Particulate medium can act as efficient damper for building internal and external vibrations. Mechanical properties of the particulate material, their shape and size distributions determine the vibration absorption characteristics of these dampers. It has been shown that an unconsolidated particulate damper act as a bandpass filter for vibration frequencies of mechanical vibrations transmitted through them. The particulate dampers can also have frequency bandgap structure, where transmitted mechanical vibration of a particular frequency is strongly attenuated by absorption and scattering. The samples of non-consolidated uniform fused-silica spherical particulates are subjected to mechanical vibrations of different frequencies and intensities to determine their absorption characteristics as a function of frequency, intensity, and direction of propagation. The effects of a particulate diameter and layer thickness on absorption characteristics are determined experimentally, at low and at high frequency ranges encountered in building vibration. The goal of this investigation is to find the required absorption characteristics for particulate damper for applications to mitigate building vibration.

10:35

4aSA8. Measurements of a wave barrier trench for isolation of rail vibration. David W. Dong and John LoVerde (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com)

Where buildings are adjacent to railroad tracks, one option for reducing levels of ground vibration is to dig a trench between the tracks and the project to act as a wave barrier. The application of this method has been rare within the United States, and consequently, practical experience is very limited. On a recent project of the authors, a wave barrier trench was constructed [Dong and LoVerde, JASA **139**, 2159 (2016)]. Measurements have been performed is various conditions both before and after construction of the trench. The vibration insertion loss was measured and compared to theory, and recommendations based on the experience are provided. This paper extends the analysis with additional information and data.

10:55

4aSA9. Isolation of residential buildings from ground vibration from railroad. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

For residential developments near railroad tracks, ground vibration due to train movement is often a concern. Although there are usually no code requirements for vibration levels, ground vibration levels are often compared to the criteria recommended by the FTA; these criteria are sometimes explicitly included as part of the Conditions of Approval or other governing documents required for the development of the site. These criteria are often exceeded when the development is in proximity to the tracks, but experience has provided limited practical applications for reducing the vibration. Some of the methods suggested include isolating the building on elastomeric bearings, digging a wave barrier trench, increasing the mass of the foundation, and stiffening the joist structure. However, these measures have been rarely employed, and little data are available on the objective effects of the mitigation, correlated with the benefit to occupants. Despite this, such methods are sometimes required on a project without analysis of efficacy or even an understanding of the anticipated result. The authors have compiled the results of recent measurements on several buildings that employed some these measures, and discuss the benefits of the mitigation measures and their application to building construction strategy.

Contributed Papers

11:15

4aSA10. Flanking sound transmission in connected panels of cross-laminated-timber at low frequencies. Blasius Buchegger, Heinz Ferk (Lab. for Bldg. Sci., Graz Univ. of Technol., Inffeldgasse 24, Graz 8010, Austria, buchegger@tugraz.at), and Martin Schanz (Inst. of Appl. Mech., Graz Univ. of Technol., Graz, Austria)

The sound transmission between adjacent rooms can be dominated by the structure-borne sound transmission in the flanking walls. A wall can be defined as a subsystem of the overall system based on two coupled rooms. These subsystems show local modes, if they are not connected to each other. In contrast, connected subsystems result in global modes showing a high response of both subsystems, if the comparable local modes have similar eigenfrequencies. The flanking sound transmission especially at low frequencies is reduced, if a generation of such global modes can be suppressed. As a consequence, the flanking sound transmission in buildings can be mitigated significantly using a proper design of the walls (geometry, material, etc.). The principles and the impact of this effect are studied in this paper using numerical models based on the finite element method. Selected constructions are implemented in real size 2D- and 3D-models of two adjacent rooms. Panels of cross-laminated-timber are used because the variable composition of the single layers of a panel allows an extensive variation of the vibration response of the single walls. The results confirm the applicability of the described effect for common practical situations.

11:30

4aSA11. Vibration isolation of large fan array air handling units in healthcare. Edward Logsdon (D. L. Adams Assoc., Inc., 1536 Ogden St., Denver, CO 80218, elogsdon@dlaa.com)

A new hospital facility constructed in Denver, CO, has all the mechanical equipment that is typically on the roof of the building located directly over all the Operating Rooms (ORs). Many of the ORs are used for procedures involving equipment found in new facilities like high powered microscopes for laser eye surgery or neurosurgery, ceiling mounted high-definition video displays and robotic surgical systems using high resolution cameras. All of which are vibration sensitive. Also, in the future the client wanted flexibility in how the operating rooms are used as the equipment and procedures will change. Because "fan array" mechanical units, produce less vibration than traditional air handling units, they were selected by the design team. The air handling units varied in size from medium with six supply and six return fans to very large with 22 supply and 16 return fans. Because of their size and location in the building, the units were loaded into the facility in multiple modular sections that were then bolted together. This required the development of a vibration isolation solution that allowed for the easy assembly of the mechanical units on site. The rational for the location of the mechanical rooms above the operating rooms will be discussed along with the vibration criteria, analysis, and details of the isolation system. This includes vibration measurements of the final results.

Session 4aSC

Speech Communication and Education in Acoustics: Second Language Speech Learning and Education

Tessa Bent, Cochair
Department of Speech and Hearing Sciences, Indiana University, 200 S. Jordan Ave., Bloomington, IN 47405

Keiichi Tajima, Cochair
Research Laboratory of Electronics, Massachusetts Institute of Technology, 36-511, 77 Massachusetts Ave., Cambridge, MA 02139

Chair's Introduction—9:00

Invited Papers

9:05

4aSC1. Predicting and explaining second language speech through individual differences: Insights from testing and extending the Second Language Linguistic Perception model. Paola Escudero (The MARCS Inst., Western Sydney Univ., Locked Bag 1797, Penrith, NSW 2751, Australia, paola.escudero@westernsydney.edu.au)

In this talk, I will present five studies aimed at demonstrating that individual native production and perception abilities predict and explain success in L2 sound perception and word recognition. These studies tested and extended the theoretical and computational framework of the Second Language Linguistic Perception (L2LP) model was tested and extended. The model's main claim states that accurate predictions for L2 success should be based on individual learners' native production and perception as well as their responsiveness to rapid perception training. In Study 1, we show how differences in how listeners produce native vowel sounds are predictive of their discrimination accuracy when perceiving L2 vowel contrasts. The predictive power of individual native and L2 vowel perception on the learning of novel L2 words are reported in Studies 2 and 3. Study 4 shows that distributional training, based on the frequency distributions of difficult non-native contrasts, leads to short and long term improvement of difficult non-native contrasts and whether this improvement generalizes the contrasts produced in different contexts and by multiples speakers. Finally, Study 5 tests whether the perception training in Study 4 has short and long term impact on the L2 word learning. A discussion of the findings and their implications for the L2LP proposal and that of other models of L2 speech will be presented.

9:25

4aSC2. Perception of American English alveolar stops and flaps by Japanese learners of English: Does allophonic variation matter? Mafuyu Kitahara (Waseda Univ./Sophia Univ., 1-6-1 Nishiwaseda, Shinjuku, Tokyo 169-8050, Japan, kirahara@waseda.jp), Keiichi Tajima (Hosei Univ., Chiyoda, Tokyo, Japan), and Kiyoko Yoneyama (Daito Bunka Univ., Itabashi, Tokyo, Japan)

A lexical decision experiment was conducted with Japanese learners of English with relatively low English proficiency to investigate whether second-language (L2) learners utilize allophonic variation when recognizing words in L2. The stimuli consisted of 36 isolated bisyllabic words containing word-medial /t/, half of which were flap-favored words (e.g., *better*, *city*) and the other half were [t]-favored words (e.g., *faster*, *custom*). All stimuli were recorded with two surface forms: /t/ as a flap (e.g., *better* with a flap) or as [t] (e.g., *better* with [t]). The stimuli were counterbalanced in the lists using a Latin Square design, so that participants only heard one of the two surface forms. The accuracy data indicated that flap-favored words pronounced with a flap (*e.g*, *city* with a flap) were recognized significantly less accurately than [t]-favored words with a flap (e.g., *faster* with a flap) and [t]-favored words with [t] (e.g., *faster* with [t]). These results suggest that Japanese learners prefer canonical forms over allophonic variants, a result which is inconsistent with Pitt *et al*.'s (2011) study with native American English listeners. The results' implications for the role of allophonic variation in L2 learning will be discussed. [Work supported by JSPS KAKENHI 26370508, 15K02492, and 16K02646.]

9:45

4aSC3. Goodness evaluation by L1 Spanish speakers of various allophones, made by human and by vocal tract models, of the Spanish phonemes /l/ and /r/. Takuya Kimura (Dept. of Spanish Lang. and Lit., Seisen Univ., 3-16-21 Higashi Gotanda, Shinagawa-ku, Tokyo 1418642, Japan, kimura@seisen-u.ac.jp) and Takayuki Arai (Dept. of Information and Commun. Sci., Sophia Univ., Tokyo, Japan)

The present study reports the results of two perceptive experiments, in which L1 speakers of Peninsular Spanish were asked to listen to a series of sound stimuli and to evaluate the goodness of each stimulus as a realization of Spanish consonant /l/ or /r/. The stimuli were made of natural human voice in the first experiment, and with a set of vocal tract models in the second experiment. The results were similar for the two experiments. The lateral approximant stimuli were highly evaluated as allophones of /l/. Some lateral

approximant stimuli were longer than others, but the duration difference caused little effect on the evaluation. The flap stimuli obtained low scores as /l/. Conversely, for the phoneme /r/, only the flap stimuli acquired high evaluations, while all the lateral approximant stimuli obtained extremely low scores. The retroflex approximant stimuli, similar to English /r/, obtained low evaluations for both /l/ and /r/ in the first experiment using a human voice, while they were evaluated slightly higher as /l/ than as /r/ in the second experiment using the vocal tract models. These results are expected to provide L2 Spanish learners with useful knowledge in their pronunciation learning.

10:05

4aSC4. Interactions between perception and production during speech sound learning. Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, mbaesebe@uoregon.edu)

It is frequently assumed that perception and production develop in tandem during non-native speech sound acquisition. However, several previous studies have suggested that simply producing tokens during training can disrupt perceptual learning (e.g., Leach & Samuel, 2009). Here, I present several experiments examining some sources of such a disruption. Further, I ask how learning in production progresses when listeners fail to learn in perception. In the first set of experiments, I examine performance on a discrimination task after training in perception alone, or training perception + production, in which listeners repeat tokens on every trial. In follow up experiments, rather than repeating the training tokens, listeners read an unrelated letter aloud between each perceptual training trial or respond to the unrelated letter with a button press, rather than reading the letter aloud. In a second set of experiments I examine how production learning progresses when learners produce tokens during training. The critical factor examined is whether listeners are able to differentiate between tokens in perception and whether this ability correlates with learning in production. Taken together, the results of these studies suggest that the relationship between perception and production is complex, especially during learning.

10:25–10:40 Break

10:40

4aSC5. Asymmetrical interpretation of English liquid consonants by Japanese speakers. Mariko Kondo (School of Int. Liberal Studies, Waseda Univ., SILS, 1-6-1, Nishi-Waseda, Shinjuku-ku, Tokyo 169-8050, Japan, mkondo@waseda.jp)

Japanese speakers have difficulty in differentiating English /l/ and /r/ because they are not contrastive. Actually, variations of both /l/ and /r/ occur in Japanese speech. The most common realization of Japanese /r/ is alveolar tap [ɾ] and all variants of both /l/ and /r/ are considered as allophones of /r/. Previously, it was found that Japanese speakers have more problems with /l/ than with /r/ in their English production. So, this study investigates why these differences occur. Analysis of Japanese speakers' mimicry speech of (a) American English and (b) English accented Japanese suggested that Japanese speakers are aware of acoustic and articulatory features of English approximant [ɹ]. Japanese speakers overused approximant [ɹ] and r-colored vowels in their mimicries of both (a) and (b). Further articulatory analysis of Japanese and English consonants showed that the English approximant [ɹ] is quite distinct from Japanese consonants, all of which lack lip rounding. The results of these studies suggest that Japanese speakers may not be able to recognize English /l/ and /r/ as separate phonemes, but that they can hear the approximant [ɹ] as it forms a new sound category, whereas they interpret /l/ as /r/.

11:00

4aSC6. Consolidation during sleep stabilizes non-native speech sound learning. Emily Myers (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Storrs Mansfield, CT 06269, emilybmyers@gmail.com)

Learning to perceive non-native speech sound categories is a challenge for many adult listeners. Even after successful learning, individuals vary in the degree to which they can successfully retain and consolidate learned perceptual categories in memory. In this talk, I describe a recent line of work from our lab showing that sleep plays a greater role in perceptual learning of non-native speech sounds than previously supposed. Specifically, we find that consolidation processes during overnight sleep protect learned phonetic information from interference from similar sounds from one's native language. I will discuss the potential sources of native language interference by reviewing a series of studies that test the low-level acoustic and abstract phonological contributions to interference. Finally, overnight gains in performance can be specifically linked to the duration and quality of sleep, suggesting a key role for sleep, per se, in the retention of non-native speech categories over multiple sessions. Taken together, this line of research suggests that individual variability in non-native speech sound learning may in part be linked to individual differences in sleep behavior and differences in the degree of pre-sleep exposure to competing native-language phonetic contrasts.

11:20

4aSC7. Neurobiological constraints on speech learning: Individual differences and plasticity. Bharath Chandrasekaran (Univ. of Texas at Austin, 1 University Station, Austin, TX 78712, bchandra@austin.utexas.edu)

A primary goal of auditory neuroscience is to understand how conspecific sounds are represented, learned, and mapped to constructs. Speech signals, conspecific to humans, are multidimensional, acoustically variable, and temporally ephemeral. A significant computational challenge in speech perception (and more broadly, audition) is categorization, that is, mapping continuous, multidimensional, and variable acoustic signals into discrete, behavioral equivalence classes. Despite the enormity of this computational challenge, native speech perception is rapid and automatic. In contrast, learning novel speech categories is effortful, and considered one of the most challenging categorization tasks for the mature brain. I will discuss three lines of ongoing research using multimodal neuroimaging, computational modeling, and behavioral training approaches: a) examine the multiple neural systems underlying successful L2 speech category learning in adulthood, b) assess sources of individual differences in L2 speech category learning, and c) design optimal, neurobiologically constrained training paradigms that reduce inter-individual differences in L2 speech category learning. These studies will provide insights into these fundamental questions: consistent with prior work on visual category learning, are multiple neural systems involved in speech categorization? What is the role of emerging expertise and individual differences in mediating neural and computational processes involved in speech category learning?

11:40

4aSC8. Individual-based accent clustering of World Englishes using the Speech Accent Archive. Nobuaki Minematsu (The Univ. of Tokyo, 7-3-1, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, mine@gavo.t.u-tokyo.ac.jp)

In a language class of English, one specific type of pronunciation is often adopted as model pronunciation and learners try to imitate that pronunciation. Even though they acquire native-like pronunciation successfully, once they go out to the real world of spoken English, they surely encounter the pronunciation diversity of English and feel the necessity to survive it. As is well-known, English pronunciation is altered variously depending on the language background of speakers, called accents. This talk describes a technical attempt of automatic and individual-based clustering of these accents using the Speech Accent Archive. This technique may be used to realize a really individual-based map of World Englishes, which can be used to introduce learners to the real state of English. For clustering, the accent gap has to be quantitatively estimated between any two speakers of the archive, where non-linguistic factors such as age and gender have to be adequately cancelled in estimation. Pronunciation structure analysis is used for cancellation and experiments show that the estimated gaps have a better correlation to IPA-based manually defined gaps than phoneme-based gaps have to the IPA-based gaps.

THURSDAY MORNING, 1 DECEMBER 2016 SOUTH PACIFIC 1, 7:55 A.M. TO 10:05 A.M.

Session 4aSPa

Signal Processing in Acoustics: Exploitation of Doppler in Acoustic Signal Processing

Paul Hursky, Cochair
HLS Research Inc., 12625 High Bluff Drive, Suite 211, San Diego, CA 92130

Shinnosuke Hirata, Cochair
Department of Systems and Control Engineering, Tokyo Institute of Technology, 2-12-1 Ookayama, S5-17, Meguro 152-8550, Japan

Chair's Introduction—7:55

Invited Papers

8:00

4aSPa1. Doppler spectral moments of multibeam echosounder data to infer the ascent velocity of methane bubbles emanating from natural seafloor seeps. Christian de Moustier (10dBx LLC, PO Box 81777, San Diego, CA 92138, cpm@ieee.org)

A multibeam echosounder operating at 500 kHz was deployed on a remotely operated vehicle to detect methane bubble streams emanating from natural seafloor seeps in 1400-m water depths. The sonar transmitted 10-ms tone bursts every 100 ms over a $7^\circ \times 120^\circ$ beam, and formed 128 receive beams regularly spaced 0.94° apart over the 120° sector. The sonar head faced upward and signals backscattered from ascending bubbles were sampled at about 6 kHz between 10 m and 50 m from the head. Doppler spectral moments were computed for each ping with the conventional data autocovariance method. For unimodal and symmetric Doppler spectra, the mean power is related to the backscattering strength of the sampled volume, the mean Doppler frequency yields the mean radial velocity of the bubbles, and the spectral width measures the mean velocity uncertainty with some indication of bubble oscillation. Mean ascent velocities of 0.2 m/s matched those obtained from concurrent optical measurements. However, beams free of bubbles showed mean descent velocities of about 1.5 cm/s, most likely due to marine snow, which could bias mean ascent velocity estimates for bubbles at greater distance from the sonar because the corresponding Doppler spectra might no longer be unimodal.

8:20

4aSPa2. A personal history of Doppler sonar development at the Marine Physical Laboratory, SIO. Rob Pinkel (MPL, SIO, Scripps Inst. of Oceanogr., 0213, San Diego, CA 92093-0213, rpinkel@ucsd.edu)

In the early 1970s, the possibility of remotely measuring oceanic and atmospheric flows was becoming appreciated. By then, techniques for information extraction from sets of discrete targets were well advanced. Inferring fluid motion from the echo off a random cloud of independent scatterers was a different issue, as illustrated by the abject failure of contemporary matched-filter technology. Conceptual as well as hardware challenges were addressed over time, leading to the variety of Doppler sonars and sodars available commercially today. In this talk, I focus on systems developed at the Marine Physical Laboratory of Scripps over the past 42 years. Both pulse-to-pulse coherent and incoherent backscatter systems are reviewed with an emphasis on real-world problems encountered. Presently, under-

exploited techniques based on sea-surface scattering or bi-static geometries are discussed. While many quality Doppler sonar manufacturers are now active, a large gap exists between what is technically possible and what is commercially available. The continuing vitality of university-based development groups that create scientifically significant (but not necessarily commercially profitable) systems is critical to advancing the state of the art.

8:40

4aSPa3. Some effects of ocean surface roughness on high and low duty cycle, broadband waveforms. Paul C. Hines (Dept. of Elec. and Comput. Eng., Dalhousie Univ., PO Box 15000, Halifax, NS B3H 4R2, Canada, phines50@gmail.com), Stefan M. Murphy (Defence R&D Canada, Dartmouth, NS, Canada), Douglas A. Abraham (CausaSci LLC, Ellicott City, MD), and Grant B. Deane (Scripps Inst. of Oceanogr., La Jolla, CA)

In this paper, data from TREX13 are used to compare the impact of the environment on High Duty Cycle (HDC) sonar and Pulsed Active Sonar (PAS). This paper presents the results of an examination of short-range single surface-reflection echoes, and longer-range target echoes from an air hose. Measurements showed that for an 18 s HDC pulse, the mean (coherent) component of the specular arrival decreased with increasing rms roughness whereas 0.5 s PAS pulse echoes showed no correlation with roughness. The standard deviations of the mean echo levels are used to examine the incoherent (scattered) component of the specular arrivals. The incoherent component of the specular arrival increased with the product of the surface correlation length and the square of the rms roughness, for both HDC and PAS, with the PAS data having a 1 dB higher standard deviation. A normal mode propagation model and a rough surface scattering model used in conjunction with a simple model that accounts for motion-induced coherence loss from the matched filter, are used to interpret the results. [Work sponsored by ONR Code 32.]

Contributed Paper

9:00

4aSPa4. Testing the feasibility of a concurrent comparison of continuous and pulsed active sonar. Stefan M. Murphy (DRDC, 9 Grove St., Dartmouth, NS B2Y 3Z7, Canada, stefan.murphy@drdc-rddc.gc.ca), Matthew Coffin (Geospectrum Technologies, Inc., Dartmouth, NS, Canada), Paul C. Hines (Elec. and Comput. Eng., Dalhousie Univ., Dartmouth, NS, Canada), and Diana F. McCammon (McCammon Acoust. Consulting, Black Rock, NS, Canada)

The performance of continuous active sonar (CAS) was compared to conventional pulsed active sonar (PAS) during the TREX13 (Target and Reverberation Experiment 2013) sea trial. The approach was to execute a one-hour CAS run followed closely by a one-hour PAS run to limit the environmental variability between runs and allow a fair comparison. This approach was possible because the sonar source and receiver were fixed to the seabed. A different approach was required for a more recent sea trial, LCAS15 (Littoral Continuous Active Sonar 2015), where the source and receiver were towed from a ship. Ship motion increases variability in sonar performance, therefore a simultaneous comparison of CAS and PAS was desired so that any motion effects were the same for both waveforms. The approach of transmitting CAS and PAS concurrently in two separate frequency bands was taken. The risk with this approach is that potential differences in propagation in the two bands could bias the comparison. A run in which equivalent CAS waveforms were transmitted simultaneously in the two bands was therefore performed to establish a baseline for the performance difference between the bands. Doppler-corrected replicas were required in all of the correlation processing due to the high Doppler sensitivity of the long-duration, wideband CAS waveforms. Preliminary results from the LCAS15 sea trial will be presented.

Invited Paper

9:15

4aSPa5. Active control of passive acoustic fields: Passive synthetic aperture/Doppler beamforming with data from an autonomous underwater vehicle. Gerald L. D'Spain, Eric Terrill, C. David Chadwell, Jerome A. Smith, and Stephen D. Lynch (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu)

Autonomous underwater vehicles (AUVs) equipped with hull-mounted arrays can actively modify received acoustic fields to optimize extraction of information. Ocean acoustic data collected by an AUV-mounted two-dimensional hydrophone array, with overall dimension one-tenth wavelength at 200-500 Hz, demonstrate aspects of this control through vehicle motion. Source localization is performed using Doppler shifts measured at a set of receiver velocities by both single elements and a physical array. Results show that a source in the presence of a 10-dB higher-level interferer having exactly the same frequency content, as measured by a stationary receiver, is properly localized and that white-noise-constrained adaptive beamforming applied to the physical aperture data in combination with Doppler beamforming provides greater spatial resolution than physical-aperture-alone beamforming and significantly lower sidelobes than single element Doppler beamforming. A new broadband beamformer that adjusts for variations in vehicle velocity on sample by sample basis is demonstrated with data collected during a high-acceleration maneuver. The importance of including the cost of energy expenditure in determining optimal vehicle motion is demonstrated through simulation, further illustrating how vehicle characteristics are an integral part of the signal/array processing structure. [Work sponsored by the Office of Naval Research.]

Contributed Papers

9:35

4aSPa6. Posterior inference of two closely spaced wave vectors for inference regarding the state of an acoustic target. Abner C. Barros (ECE Dept., Univ. of Massachusetts Dartmouth, Darmouth, MA), David Anchieta (Elec. and Comput., Universidade Federal do Pará, Belém, Brazil), and Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, pgendron@umassd.edu)

A bistatic sonar arrangement is employed to infer the depth, speed, and range of an oncoming submerged acoustic target from a continuous wave transmission. Surface and direct interacting paths associated with closely spaced angles and frequencies from the returned acoustic wave fronts are resolved. A Gibbs sampler is employed to construct the posterior joint density of all parameters taking full advantage of the analytic tractability of the amplitude and phase of the wave fronts while the ordered wave vectors, their angles and frequencies are sampled numerically by two-dimensional quantile function. The inferred density of target depth, range, and speed is accomplished by constructing a numerical inverse-transformation of the forward propagation model. Simulations are used to demonstrate the potential of sub-Rayleigh resolution accuracy at less than -10dB received SNR with an 9 element vertical array.

9:50

4aSPa7. Processing of aircraft signatures with significant Doppler, simultaneously observed on microphones and distributed underwater sensors. Paul Hursky (HLS Res. Inc., 12625 High Bluff Dr., Ste. 211, San Diego, CA 92130, paul.hursky@hlsresearch.com)

An experiment was performed with a source mounted on an aircraft with a microphone array rigidly mounted on a research vessel, and with sonobuoys and a vertical line array deployed in the ocean underneath. Programmed waveforms, including tones, linearly frequency modulated tones, and m-sequences, were transmitted from the airplane. The Doppler in this configuration was substantial, due to the sound speed in air being 1/5 that of water, with much of the acoustic path passing through the air. We will present results of matched filtering the transmitted waveforms, using various combinations of observed signals, including using replicas augmented by modeling, to compensate for the Doppler in order to enhance the coherent integration gain in this configuration.

THURSDAY MORNING, 1 DECEMBER 2016 SOUTH PACIFIC 1, 10:20 A.M. TO 11:55 A.M.

Session 4aSPb

Signal Processing in Acoustics: High Resolution Imaging Sonars Including Real Aperture, Synthetic Aperture, and Tomographic Sonars I

Brian G. Ferguson, Cochair
DSTO, PO Box 44, Pyrmont, NSW 2009, Australia

Timothy Marston, Cochair
APL-UW, 1013 NE 40th Street, Seattle, WA 98105

Chair's Introduction—10:20

Invited Papers

10:25

4aSPb1. High resolution systems: The resolution argument. Yan Pailhas and Yvan Petillot (Heriot-Watt Univ., Riccardo Campus, School of EPS, Edinburgh EH14 4AS, United Kingdom, Y.Pailhas@hw.ac.uk)

Resolution comes from the Latin word *resolvo*, which literally means "untie a knot." The meaning evolved, in optics, toward the capability to distinguish two scattering points. For the MCM (Mine Counter Measure) problem, the introduction of synthetic aperture systems, and the resulting drastic increase in resolution, has been a game changer, as the useful information shifts from shadows to highlights. If SAS resolution has been well studied, new SAS reacquisition patterns have emerged, in particular circular acquisitions known as CSAS. In this talk, we will derive the theoretical resolution of such systems based on a PSF (point spread function) energy leakage interpretation rather than the traditional Rayleigh resolution criteria. We will compare the CSAS resolution capability with the newly developed MIMO paradigm. The statistical framework of MIMO systems greatly differs from the deterministic approach of synthetic aperture systems, and offers certain advantages in term of resolution. All these coherent systems approach the imaging problem without any priors, other than the presence of scatterers. Going back to the original meaning of *resolvo* however, we will show that, by introducing priors such as target shape, wideband sonars shift the analysis into the phase domain and are able to outperform imagery.

10:45

4aSPb2. Moving beyond side-looking imaging with synthetic aperture sonar. James Prater and Jonathan King (NSWC PCD, 110 Vernon Ave., Panama City, FL 32407, jonathan.l.king1@navy.mil)

The desire to provide long-range high-resolution side-scanning sonar imagery necessitates the use of lower frequencies to overcome the relative high attenuation of high frequency sound waves and long apertures in order to achieve high resolution. This is typically accomplished through the application of synthetic apertures which collect wideband widebeam sonar data over time and post-process the data to produce high resolution imagery. While many of these systems were initially designed with high resolution long range imagery in mind, the wideband widebeam and temporal nature of the data provides the potential for signal processing and analysis to produce information about the imaged area beyond image intensity. This paper will discuss recent advances in SAS signal processing that exploit the wideband, widebeam, and temporal features of unbeamformed SAS data to extract additional information about the imaged area. Specifically, techniques that exploit aspect dependence (from widebeam data), frequency dependence (from wideband data), and temporal phenomena will be discussed.

11:05

4aSPb3. Effect of low complexity adaptive beamforming on seafloor backscatter statistics. Tor I. Lønmo (Dept. of Informatics, Univ. of Oslo, P.O.Box 111, Horten 3191, Norway, toribi@ifi.uio.no) and Anthony Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

In contrast with the common delay-and-sum (DAS) beamformer, adaptive beamformers can be used to increase the resolution of acoustic measurement systems without sacrificing sidelobe levels. The non-linear nature of adaptive beamforming methods may cause changes to measured scattered amplitude distributions. For example, large differences in speckle statistics have been observed in some cases when using the adaptive minimum variance distortionless response (MVDR) beamformer in medical ultrasound, unless preventative measures were implemented. Low complexity adaptive (LCA) beamforming techniques have shown similar, but smaller, differences. Recently, experiments have been conducted in the Oslofjord, Norway, to determine whether low-complexity adaptive (LCA) beamforming techniques can improve the performance of bathymetric echosounders. When using LCA with these types of sonar systems, it is important to understand any changes to the backscatter statistics, as these changes may affect, e.g., seafloor classification methods. In this talk we will present comparisons between backscatter statistics estimated using a weighted delay-and-sum beamformer and those estimated using a low complexity adaptive beamformer. We will discuss any differences seen, possible mechanisms causing these differences and potential remedies. [Research supported by The Norwegian Research Council and Kongsberg Maritime. The first author is employed in Kongsberg Maritime.]

Contributed Papers

11:25

4aSPb4. Paradigm shift of underwater echo sounding technology I—Principle and algorithm. Ikuo Matsuo (Dept. of Information Sci., Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, matsuo@mail.tohoku-gakuin.ac.jp) and Toyoki Sasakura (FUSION INC., Tokyo, Japan)

Underwater echo sounding technology was developed 65 years ago and has been applied in various systems such as fishfinder, bathymetry, sonar, side scanning sonar, multi beam echo sounding system, and synthetic aperture sonar. Here, we proudly suggest a new concept that may change the history of underwater echo sounding technology, the "Paradigm Shift." In the conventional system, the transmitting signal is transmitted after receiving the target signal, e.g., a bottom echo. Therefore, the transmission intervals are always longer than the time of the round trip distance to the target divided by the underwater sound velocity. On the other hand, by adapting our new "Paradigm Shift" into the system, the transmission intervals can be decided without depending on a target and it will be possible to conduct a bathymetry survey that transmits 100 times per second. The system was tested and confirmed in the actual sea trial. This new concept, the "Paradigm-Shift-Echosounder" (PSE), utilizes the 7th order and 127 kinds of Gold code sequence signals. Different code of Gold code signals will be transmitted in each transmission, and receiving signals will be identified by the correlation of the 127 kinds of Gold code sequences of different transmission signals.

11:40

4aSPb5. Paradigm shift of underwater echo sounding technology II—Experiment and actual sea trial. Toyoki Sasakura (FUSION INC., 1-1-1-806, Daiba, Minatoku, Tokyo 1350091, Japan, sasakura@fusion-jp.biz) and Ikuo Matsuo (Dept. of Information Sci., Tohoku Gakuin Univ., Sendai, Japan)

The newly suggested "Paradigm-Shift-Echosounder" (PSE) for bathymetry survey was tested in a sea trial experiment using a prototype. Experiment was conducted at the depth of 30-80 m using 200 kHz signals transmitted 100 times per second and succeeded to obtain continuous bottom echo data. We also conducted a side scanning sonar experiment using 88 kHz signals, at a site with a depth more than 500 m and succeeded to obtain the echo of a 2 m x 5 m artificial reef moored at the depth of 50 m. Correlation processing was done and we managed to reconstruct a high-resolution echogram of the artificial reef. The prototype used in the trial experiments utilizes the 7th order Gold code sequence. Four cycles of carrier signals as 1 bit, the transmission signals are all phase-modulated. The pulse length of transmission signal is 2.54 ms for 200 kHz. As there are 127 Gold codes, transmission signals of the same code are repeated every 1.28 seconds. The receiving signals are then processed by the correlator, with a shift register length of 4064 stages by 8-times sampling frequency for carrier frequency. There are 127 kinds of replicas of correlator, and the numbers of "Exclusive OR + Summation" circuits are also 127.

Session 4aUW

Underwater Acoustics, Signal Processing in Acoustics, and Acoustical Oceanography: Harbor Acoustics I

Peter J. Stein, Cochair
Scientific Solutions, Inc., 99 Perimeter Road, Nashua, NH 03063

David L. Bradley, Cochair
Penn State University, PO Box 30, State College, PA 16870

Chair's Introduction—8:15

Invited Papers

8:25

4aUW1. Physical oceanographic modeling of the inner Continental Shelf. Alan F. Blumberg (Stevens Inst. of Technol., Stevens Inst. of Technol., 1 Castle Point, Hoboken, NJ 07030, ablumberg@stevens.edu)

The nowcast and forecast of the marine environment in the shallow waters (0—20m depth) adjacent to the coast, the inner shelf, is important for the development of acoustic surveillance systems. These waters are among the most challenging marine environments in the world because they are subject to the combined geometrical constraints of irregular coastlines and highly variable bathymetry, and are forced both by a complex array of tidal, wind, and buoyancy forces on a broad range of space/time scales. Their proximity to the deeper waters flowing offshore introduces another array of complex sets of forcing. The resulting circulation patterns include both persistent and time-variable fronts, intense currents with strong spatial (offshore and/or vertical) dependence, waves, internally generated variability, large horizontal water mass contrasts, strong vertical stratification, and regions of intense turbulent mixing in both surface and bottom boundary layers. Frequently, the interplay between these multiple forcing mechanisms, geographic detail, stratification, and nonlinear dynamics, is significant, and this demands that a physically sophisticated coastal ocean model be capable of representing a comprehensive suite of dynamical processes. Drawing on the operational shallow water New York Harbor Observing and Prediction System (NYHOPS), we illustrate, by example, the breadth of dynamical processes that influence near coastal ocean circulation.

8:45

4aUW2. Three-dimensional acoustic modeling of a complex harbor environment. Michael B. Porter (Heat, Light, and Sound Res., 3366 N. Torrey Pines Ct., Ste. 310, La Jolla, CA 92037, mikeporter@hlsresearch.com), Laurel J. Henderson (Heat, Light, and Sound Res., San Diego, CA), Timothy F. Duda, and Arthur E. Newhall (Woods Hole Oceanographic Inst., Woods Hole, MA)

Typically, we model sound propagation in the ocean viewing it as an "open" environment where the sound propagates outwards from the source. Of course, targets are reflectors, and other features such as islands and seamounts have been of interest as false targets. However, the standard acoustic models based on rays, normal modes, spectral integrals, and parabolic equations generally consider outgoing energy confined to a single bearing line. In contrast, constrained or partially closed environments such as harbors introduce special issues. Obviously, the sound can be reflected back by seawalls and ship hulls, to mention just a couple of features of harbors environments. Considering coastal environments more broadly, sound reflections also may be caused by wharves, docks, piers, boat landings, quays, floats, groins, and jetties as well as natural features such as submarine canyons. In short, we have sound refracted in the latitude/longitude plane and reflected by a variety of features. In terms of computational ocean acoustics there are few options for modeling such problems. We discuss a comprehensive 3D beam tracing approach implemented in a model called BELLHOP3D that can handle 3D effects including complete reversal of the sound. It is also capable of handling narrowband waveforms (tonals) as well as full broadband timeseries. We will demonstrate and test the model using an example location that is expected to become the site of a future sea test.

9:05

4aUW3. Eigenray complexity in harbor environments. Ruth E. Keenan (Appl. Res. Laboratory/UT, 10000 Burnet Rd., Austin, TX 78758, rkeenan@arlut.utexas.edu)

Acoustic eigenray complexity patterns will be demonstrated in a simulated harbor environment. Using bistatic surface and bottom scattering strength models, reverberation levels will be estimated for the same environmental parameterization. Eigenrays are the basis of any acoustic simulation, especially an active acoustics simulation, that requires high fidelity arrival time and arrival angle information. The sensitivity of cross harbor acoustic links, ranges 1 to 5 km, will be examined for their response to fresh water intrusions (as evidenced by sharp temperature and salinity changes due to river outflow and rain events) as well as the diurnal surface heating and wind effects. The optimal acoustic frequency will be examined for its' sensitivity to the temperature and salinity changes which can significantly influence the volume attenuation. Harbors are typically very shallow water environments and the effect of the tidal cycle on the propagation will be examined as well. The spatial bathymetry and sediment composition variability over the acoustic links will also be examined.

9:25

4aUW4. Underwater sound propagation and scattering in Ports and Harbors. Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

Ports and Harbors are crucial life lines in our economy and commerce. To ensure safe and efficient utilization of these important facilities, a thorough understanding of underwater sound propagation and scattering in very shallow water regions is critical. These regions however, present a challenging waveguide for underwater acoustics. Weather the goal is active acoustics (such as underwater communication) or passive acoustics (such as source detection and localization), detailed understanding of channel response function is necessary in order to develop and apply proper security measures. Several studies in recent years have addressed interesting results in acoustic signal propagation and scattering. These include effects of sound speed profile in water column, sea surface dynamic behavior, and seafloor attenuation. Signal coherence, and spatial and temporal variabilities due to environmental parameters have all been examined and to some extent modeled. Deterministic features due to interaction of broadband acoustic signals with the dynamic environment are shown. These results have been incorporated in development of robust underwater communications that can also be utilized for Harbor security implementation. In this paper, we present results of several recent experiments and modeling using Ray and Parabolic Equation approaches to understand this problem. [Work supported by ONR 322 OA.]

9:45

4aUW5. Building a practical model for acoustic harbor surveillance. David R. Dall'Osto, Eric I. Thorsos, and D. J. Tang (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu)

The ability to simulate acoustic field in very shallow and confined environments is essential for both design and for successful operation of harbor protection sonar systems. In order to predict the acoustic field, which includes scattering from targets and the environment, a model has been developed to simulate active sonar operations. This paper uses this model to investigate the effects of environmental features common to most harbors, including sloped bathymetry from dredged areas, a variable sound-speed profile, and the effect of tides. These features are discussed in terms of their effect on detection of targets of interest, and the model is used to predict the best remedy for the adverse features. Guidelines for adequate planning of a harbor security sonar system are provided in response to specific harbor features. These include what sonar signals, beam-patterns, and multi-static deployment geometry will provide optimal detection.

Contributed Paper

10:05

4aUW6. 3D modeling of low-to-mid-frequency acoustic propagation through an estuarine salt wedge. D. Benjamin Reeder (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943, dbreeder@nps.edu) and Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

An estuary is an example of a constrained environment which often hosts a salt wedge, the stratification of which is a function of the tide's range and speed of advance, river discharge volumetric flow rate, and river mouth morphology. A field experiment was carried out in the Columbia River to test the hypothesis that the estuarine salt wedge is acoustically observable in terms of low-to-mid-frequency acoustic propagation. Linear frequency-modulated (LFM) acoustic signals in the 500-2000 Hz band were collected during the advance and retreat of the salt wedge. Data-model comparisons demonstrate the degree to which acoustic propagation in this constrained environment is controlled by horizontal refraction and out-of-plan scattering by bedforms along the acoustic transect.

10:20–10:35 Break

Invited Papers

10:35

4aUW7. Bioacoustics of fishes (and invertebrates) in harbors. Arthur N. Popper (Univ. of Maryland, Biology/Psych. Bldg., College Park, MD 20742, apopper@umd.edu)

Harbors can be noisy places. While some sounds are natural, there is often a cacophony of transient and continuous man-made sound from boating, shipping, construction, bridge traffic, near-by roadways, tunnels, etc. The question is whether these sounds potentially interfere with the behavior of resident and transient fish and invertebrate populations. The greatest effect may result from acoustic masking which would shorten the distances over which animals can communicate and detect sounds of biological importance such as from potential predators and prey. In order to understand potential effects of man-made sounds on harbor animals, much needs to be known about such sounds. However, the nature and extent of man-made sound has rarely been measured in harbors. Even when studied, focus has often been either on frequencies outside of the hearing range of the fishes and invertebrates and/or done for very short time periods so that a real sample of the daily and seasonal cycle of sounds is not examined. Finally, studies are usually limited to sound pressure, whereas most fishes and invertebrate primarily detect particle motion. This paper will discuss the man-made sounds of harbors and relate these to potential effects on the aquatic inhabitants.

10:55

4aUW8. Marine mammal hearing and the effects of noise. Brandon L. Southall (SEA, Inc., 9099 Soquel Dr., Ste. 8, Aptos, CA 95003, Brandon.Southall@sea-inc.net)

Marine mammals use sound for critical life functions through passive listening and, in some species, active sonar. Major recent advances have been made in understanding what sounds they make, how they hear, and how noise affects their hearing, behavior, and physiology. While data are lacking for whole groups, notably baleen whales, direct hearing measurements are available for most taxa. Some species, such as odontocete cetaceans, hear extremely broad frequency ranges, well into ultrasonic biosonar frequencies. Pinnipeds have more limited frequency ranges but are adapted for aerial and underwater hearing. Indirect data suggest whales use and are more sensitive to low frequencies. For most species, quite high received sound levels are required to temporarily or permanently affect hearing. Direct experimental measurements of behavioral responses have revealed strong species and context dependencies, with factors such as proximity, novelty, and spatial relationships influencing response probability as well as relative loudness. For active surveillance sonar systems operating in the 30-200 kHz range, whales and otariid pinnipeds are of much less concern in terms of audibility and potential behavioral disturbance than phocid pinniped and odontocete cetaceans, notably porpoises. Hearing loss or other injuries from such systems are quite unlikely in almost all practical scenarios.

11:15

4aUW9. Characteristics of surface scattering of very high frequency acoustic signals. James Preisig (JPAnalytics LLC, 638 Brick Kiln Rd., Falmouth, MA 02540, jpreisig@jpanalytics.com) and Grant Deane (Scripps Inst. of Oceanogr., La Jolla, CA)

Many applications of acoustics in harbors are characterized by relatively shallow sources and receivers and a sound speed profile that does not strongly refract signals away from the surface. Therefore, surface scattering is an important factor in determining the characteristics of propagating signals in harbor environments. In addition, applications often involve relatively short propagation distances so very high frequency (VHF) acoustic signals, defined here to be signals above 100 kHz in frequency, can be important components of acoustic systems in harbor environments. These VHF signals have very short wavelengths (approximately 2.7 mm at a frequency of 550 kHz) so surface waves as small as capillary waves can have a significant impact on the surface scattered signals. Thus, even in light wind and wave conditions, the surface scattering of VHF can be highly dynamic. A test of the surface scattering of acoustic signals between 350 kHz and 750 kHz by wind-driven waves was conducted in a wave tank with the received signals being recorded at a vertical line array. This talk will present the analysis of the characteristics of the surface scattered signals including their spatial structure and Doppler spreading and the periodicity of their temporal fluctuations.

11:35

4aUW10. Acoustic monitoring of the bottom characteristics of a harbor. Christian de Moustier (10dBx LLC, PO Box 81777, San Diego, CA 92138, cpm@ieee.org)

Acoustic monitoring of the underwater environment typically found in a harbor requires knowledge of the spatial and temporal variability of the noise, the water properties, the currents, and the bottom characteristics. For safety of navigation, repeat bottom surveys must be performed at regular intervals to monitor changes in water depth along navigation channels due to accumulation or depletion of sediments, and to detect potential hazards on or near the bottom. Imaging side-looking sonars, particularly those with synthetic aperture capabilities, are most effective for hazard detection but their performance in waters 5-10 m deep can be hampered by multipath and by changes in water properties at the scale of the scene being imaged. Their inherent spatial filtering characteristics make multibeam echosounders effective tools for synchronous bathymetry and acoustic backscatter imagery of the water column and the bottom. In harbor surveys, such echosounders achieve decimetric spatial resolution by operating at several hundred kilohertz, but their measurements are sensitive to spatial and temporal changes in water properties at the scale of the survey swath, potentially reducing lateral coverage. The corresponding variations in acoustic absorption affect acoustic backscatter imagery and yield increased uncertainty in the outcome of a repeat harbor survey.

Session 4pAAa

Architectural Acoustics: Assorted Topics in Architectural Acoustics II

Niels W. Adelman-Larsen, Cochair
Flex Acoustics, Diplomvej 377, Kgs. Lyngby 2800, Denmark

Reiji Tomiku, Cochair
Department of Architecture, Oita University, Dannoharu 700, Oita 8701151, Japan

Contributed Papers

1:00

4pAAa1. Acoustics for amplified music and a new, variable acoustics technology. Niels W. Adelman-Larsen (Flex Acoust., Diplomvej 377, Kgs. Lyngby 2800, Denmark, nwl@flexac.com)

Surveys among professional musicians and sound engineers reveal that a long reverberation time at low frequencies in halls during concerts of reinforced music such as pop and rock is a common cause for an unacceptable sounding event. Mid- and high-frequency sound is seldom a reason for lack of clarity and definition due to a 6 times higher absorption by audience compared to low frequencies, and a higher directivity of speakers at these frequencies. Lower frequency sounds are, within the genre of popular music, rhythmically very active and loud, and a long reverberation leads to a situation where the various notes and sounds including vocals cannot be clearly distinguished. This reverberant bass sound rumble often partially masks even the direct higher pitched sounds. A new technology of inflatable, thin plastic membranes presents a solution to this challenge of needed low-frequency control. It is equally suitable for multipurpose concert halls that need to adjust their acoustics by the push of a button and for halls and arenas that only occasionally present amplified music and need to be treated just for the event. The technology, permanently mounted, is being projected in various concert halls around the world and is being installed in the new Dubai Opera and the Sheik Jaber Al Ahmad Cultural Center, Kuwait during spring 2016. This paper presents the authors' research of also large venues as well as the technology showing applications including on/off measurements of reverberation time versus frequency.

1:15

4pAAa2. Proposed design schemes for vineyard concert halls with subjective test of solo singing. Weihwa Chiang, Yi-Run Chen, and Melisa Rosiana Tanadi (College of Design, National Taiwan Univ. of Sci. and Technol., 43 Keelung Rd., Section 4, Taipei 106, Taiwan, whch@mail.ntust.edu.tw)

Although a vineyard hall has both acoustical and architectural merits associated with its short source-receiver distance, attention to acoustical design needs to be placed upon the terrace seating that primarily governs the energy distribution in such a hall, in particular when listening to the vocal sound where both acoustic and visual deprivation exists. Design schemes were developed sequentially with an emphasis on seating division and evaluated by computer modeling using Odeon 9.2 acoustic software for a hall with approximately 1/4 of the seats surrounding the stage. The effect of solo singing was evaluations by listeners representing musicians and the general public based on pair comparison. Anechoic soprano and baritone singing segments with calibration information were recorded. Side terraces, preferably stepped back or splayed toward both ends of the hall, are essential for controlling the overall energy distribution. 3-dB high-frequency energy compensation can be provided for the terrace seats surrounding the stage by introducing a frontal terrace and terraces near the corners of the hall. The improvement in objective measures was verified by subjective evaluation where the perceived quality at a terrace seat surrounding the stage was not significantly less than the ones facing the stage.

1:30

4pAAa3. Electronic shell-Improvement of room acoustics without orchestra shell utilizing active field control. Takayuki Watanabe and Hideo Miyazaki (Yamaha Corp., 10-1 Nakazawa-cho, Nakaku, Hamamatsu 430-8650, Japan, takayuki.watanabe@music.yamaha.com)

This paper introduces an example of Electronic Shell acoustic enhancement system that was installed in a multi-purpose hall without an orchestra shell. The system is based on the concept of Active Field Control using electro-acoustic means. The three objectives of this system were 1) the enhancement of early reflection for performers, 2) the extension of the reverberation time and the enhancement of the total sound energy on stage, and 3) the enhancement of early reflection in the audience area. The application of this system showed an improvement of about 1 to 2 dB in STearly and more than 2 dB in G in the audience area, which is equivalent or better performance than simple mobile type orchestra shell. Regarding the spatial sound structure and the time domain characteristics between simple mobile type shell and the Electronic Shell system, there are differences since the mobile type shell mainly takes care of lateral reflections; on the other hand, the Electronic Shell system takes care of overhead reflections, and it cannot provide some early reflections after the direct sound arrives. It was clarified the relationship between the differences in physical features and the subjective improvement of the musical performance using a temporal system installed in a rehearsal hall.

1:45

4pAAa4. Stadia acoustic atmosphere and spectator experience: Quantifying crowd noise with architectural form. Elizabeth Valmont, Matthew Wilkinson, and Ashwin Thomas (Acoust., Arup, 12777 West Jefferson Boulevard, Bldg. D, Ste. 100, Los Angeles, CA 90066, elizabeth.valmont@arup.com)

Very little reliable data exist on noise levels and associated crowd atmosphere in stadia. Discussed frequently in the media, several stadia are often portrayed as having the "best" and/or "noisiest" atmosphere. However, there is no accurate basis or measurement for these claims. This poses challenges when designing new stadiums with the intent to improve spectator experience. Architectural elements such as overhangs and vertical sound reflecting surfaces are well known to redirect acoustic energy and improve the overall noise level. However, the direct impact of these elements on crowd behavior and overall atmosphere is not well documented and limits quantifying drivers for new designs. Noise level data were gathered in this study from various stadia and correlated to architectural form using acoustic modeling. The paper details the data gathered at football games played at Autzen Stadium (partial roof overhang) and the Los Angeles Memorial

Coliseum (open bowl architecture). The measured statistical data are analyzed and linked to the results of computer acoustic models of each stadium. Using the known dimensions of each stadium, the effect of architectural elements on the direct and reverberant noise level in the stands is presented.

2:00

4pAAa5. Diffusion unwrapped, new testing that reveals real data. Richard L. Lenz (RealAcoustix LLC, 2637 N. Washington Blvd., #125, N. Ogden, UT 84414, RL@RealAcoustix.com)

The concept of acoustic diffusion and devices that create diffuse properties has been around for well over 40 years. While there have been a number of test methods that have purported to show diffusion from these devices, the evidence of this is questionable at best. The author, along with others, has developed a new test method that can differentiate between specular and diffuse reflections as well as show many other characteristics desirable for acousticians and designers. This test method, being developed in coordination with ASTM, allows the user to see many different components of diffusion including polar responses, frequency response, attenuation, and phase characteristics, a key component of understanding a diffuse reflections. At the same time, the test method also shows the character of specular reflections, which difference is quite clear in the output data. The paper will also show how devices, from quadratic residue diffusers to geometric shapes, do not necessarily have the frequency response or diffuse reflections that we have assumed for many years. Many different examples will be presented as well as the different data outputs required to see how a diffuser actually performs.

2:15

4pAAa6. Diffusion: Better ways of measuring it, displaying the data and using it in room simulations. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

This paper will present and discuss diffusion and the many problems with trying to measure it, displaying the data and using this data in ways that allow us to better predict room responses. We will quickly discuss prior standards, their strengths and weaknesses and how a new proposed ASTM standard will correct those weaknesses. We will show that methods used in other areas of acoustics and electro-acoustics can be used to measure diffusion and display it in ways that allow directivity, magnitude, and phase to be displayed. These data sets can then be used in simulation programs, such as EASE. Odeon and CATT Acoustics, to better predict room responses for use in the design of rooms. We will present examples of those uses and the advantages in these proposals.

2:30

4pAAa7. Designing of variable Schroeder diffusers using digital architecture techniques. Hassan Azad (Architecture, Univ. of Florida, 3527 SW, 20th Ave., 1132B, Gainesville, FL 32607, h.azad@ufl.edu) and Gary Siebein (Siebein Assoc., Inc., Gainesville, FL)

Schroeder diffusers consist of a series of wells of the same width and different depths. The wells are separated by thin fins. The depths of the wells are determined by a mathematical number sequence, such as the quadratic residue sequence. The sequence number for the n^{th} well, S_n, is given by[1]: $S_n = n^2$modulo N where modulo indicates the least non-negative remainder and is often written as mod for short. N is the number generator which in this case is also a prime and the number of wells per period. The well width follow the simple equation as follow in which λ_{min} is the minimum wavelength before cross-modes in the wells appear[1]: $W = \lambda_{min} / 2$ In this paper it is intended to build a physical model of a N = 7 diffuser and make the depth of each well separately variable by means of digital architecture tools. The tools which is referred to would be Arduino kit micro controller and 7 stepper motors and 7 drivers. The idea behind this practice is twofold. First is to come up with a variable diffuser which works in several (preferably 5) design frequencies and hence their integer multiples. The user can make this adjustment via a simple computer program. Second, to try other mathematical number sequence and observe the diffusion characteristics of new products using the analyses of acoustic measurement results. 1. Acoustic Absorbers and Diffusers Theory, Design and Application, T. J. Cox and P. D'Antonio, 2^{nd} ed.

2:45

4pAAa8. Finite element sound field analysis for investigation on measurement mechanism of reverberation time in non-diffuse sound field. Reiji Tomiku, Toru Otsuru (Dept. of Architecture, Oita Univ., Dannoharu 700, Oita 8701151, Japan, tomiku-reiji@oita-u.ac.jp), Noriko Okamoto (Dept. of Architecture, The Univ. of Kitakyushu, Oita, Japan), Takeshi Okuzono (Dept. of Architecture, Kobe Univ., Oita, Japan), and Youtarou Kimura (Dept. of Architecture, Oita Univ., Oita, Japan)

As well known, the reverberation time is essential to evaluate room acoustics; however, it is not so easy to predict in a practical room where the assumption of diffuse sound field is not satisfied. On the other hand, the numerical analyses based on the wave equation have been intensively used to explore many kinds of acoustic problems. In this study, 15 kinds of rooms with different absorption surfaces are analyzed by time domain finite element method. This study shows effectiveness of the analysis for investigation on measurement mechanism of the reverberation time of sound field in rooms where the assumption of diffuse sound field is not satisfied. To evaluate a sound field in a room for the measurement, the ratio of incident sound energy to a part of boundary in those to all boundary of a sound field in a room is calculated from results of the time domain finite element analysis. At first, the calculation method of the ratio of incident sound energy to a part of boundary in those to all boundary of a sound field in a room is described, and the ratios are calculated in the 15 kinds of rooms at each frequency. Finally, relationship among the ratios, mean absorption coefficients, reverberation times calculated from the results of TDFE analysis and those from Sabin's formula is investigated.

3:00

4pAAa9. Spatialized sound reproduction for telematic music performances in an immersive virtual environment. Samuel Chabot (Rensselaer Polytechnic Inst., 40 3rd St., Troy, NY 12180, chabos2@rpi.edu)

Telematic performances connect musicians and artists at remote locations to form a single cohesive piece. As these performances become more ubiquitous as more people have access to very high-speed Internet connections, a variety of new technologies will enable the artists and musicians to create brand new styles of works. The development of the immersive virtual environment, including Rensselaer Polytechnic Institute's own Collaborative-Research Augmented Immersive Virtual Environment Laboratory, sets the stage for these original pieces. The ability to properly spatialize sound within these environments is important for having a complete set of tools. This project uses a local installation to exemplify the techniques and protocols that make this possible. Using the visual coding environment MaxMSP as a receiving client, patches are created to parse incoming commands and coordinate information for engaging sound sources. Their spatialization is done in conjunction with the Virtual Microphone Control system, which is then mapped to loudspeakers through a patch portable to various immersive environment setups.

3:15

4pAAa10. A simplified approach to virtual reality and auralizations for architectural acoustics. Christopher Buckley (Omny Acoust., LLC, PO Box 757, Port Austin, MI 48467, info@omnyacoustics.com) and Giuliano Bernardi (none, Leuven, Belgium)

The use of auralizations to convey important architectural acoustics design decisions is well known, but the use of accurate auralizations paired with virtual reality (VR) are typically limited to new, large-budget constructions. Smaller-budget projects often cannot afford the expensive software packages necessary to perform such auralizations, or are limited by design time or computational resources. In order to provide accurate VR-enabled auralizations for a larger variety of projects, this paper demonstrates a simplified method which can be used with a pre-calibrated combination of a smartphone and headphones. The presented method allows for modeling room-to-room sound transmission in addition to 3D spatialization cues from early reflections and reverberation. The room-to-room sound transmission is handled by simple filtering operations, which simulate the transmission loss of the demising partition, while the 3D spatialization cues are generated from one of the many freely available spatialization engines such as the Oculus Native Spatializer or Google VR Spatial Audio. This simplified method does not intend to replace existing more robust solutions, but instead allows for a more wide-spread usage of VR-enabled auralizations in cases where a high performance computer or detailed 3D model are not available.

Session 4pAAb

Architectural Acoustics: It's All About the Details

Shane J. Kanter, Cochair
Threshold Acoustics, 53 W. Jackson Blvd., Suite 815, Chicago, IL 60604

John T. Strong, Cochair
Threshold Acoustics, 53 W. Jackson Blvd., Suite 815, Chicago, IL 60604

Kimihiro Sakagami, Cochair
Dept. of Architecture, Grad. School of Engrg., Kobe University, Rokkodai 1-1, Nada, Kobe 657-8501, Japan

Motoki Yairi, Cochair
Kajima Technical Research Institute, Chofu, Tokyo 182-0036, Japan

Chair's Introduction—3:45

Invited Papers

3:50

4pAAb1. Numerical study on the absorption performance of MPP sound absorber installed in small rectangular rooms. Takeshi Okuzono and Kimihiro Sakagami (Architecture, Kobe Univ., 1-1 Rokkodai-cho, Kobe, Hyogo 657-8501, Japan, okuzono@port.kobe-u.ac.jp)

Sound absorbers using microperforated panel (MPP) are one of the most promising alternatives of the next-generation sound absorbing materials, thanks to the superior material performances in durability, weatherability, recyclability, and flexibility of design, as well as the attractive broadband sound absorption characteristics. Many previous researches focus on the development of absorbers itself. However, in order to bring out the absorption performance sufficiently or to use it appropriately in room acoustics applications, it is important to understand the absorption effect of MPP absorbers on sound fields in rooms. For the purpose, recently, highly accurate wave-based acoustics simulation techniques such as FEM can be used to analyze sound fields in practical sized rooms with MPP absorbers. This paper presents the absorption performance of MPP absorbers installed in a small rectangular room of 91 m^3 using 3D-FEM in frequency domain. Both the steady-state and the unsteady-state sound fields are calculated up to 1 kHz under the various absorber locations. The unsteady-state sound fields are calculated using inverse Fourier transform of transfer functions of rooms. The results are compared to explore the optimum sound absorber locations.

4:10

4pAAb2. Effect of the mass-spring resonance of a slab covered with carpets on the sound insulation performance. Motoki Yairi, Atsuo Minemura, and Takashi Koga (Kajima Tech. Res. Inst., Chofu, Tokyo 182-0036, Japan, yairi@kajima.com)

It is well known that the overall sound insulation performance between two adjacent rooms in actual buildings is not so much increased even if a high spec wall is used for the partition because of several flanking sound transmission paths. In recent years, the phenomena that a slab covered with carpets increases the sound transmission through the floor have been found. This is caused by the resonance with the mass of the carpets and the spring of those underlying materials, and consequently the sound insulation performance significantly decreases in the mid-low frequency range. The resonance frequency can be shifted to higher frequencies using formed plastic sheets of higher stiffness for the underlying materials. Although this would usually be one of countermeasures to increase the overall sound insulation performance, it is not a fundamental solution for noise control engineering. In the present work, in order to control the mass-spring resonance itself, a method of focusing on permeability of the carpets is proposed. The resonance is expected to be damped with the flow resistance, which is caused by setting an appropriate permeability. It is theoretically investigated using a simple model, and an experimental validation is also carried out.

4:30

4pAAb3. Essential details in performing art facilities' practice rooms. Jennifer Nelson Smid (Threshold Acoust., 141 W Jackson Blvd, Ste. 2080, Chicago, IL 60604, jnelson@thresholdacoustics.com)

Limited space and multi-use programming often leaves practice rooms working for several instrumental and vocal groups within performing arts facilities. During programming, communication about expected performance of the space and adjacencies, as well as design team collaboration will help to keep the project in scope and on budget. Special details for multiple trades are required to design a well isolated and multi-functioning practice room for various quantities and ranges of instruments and voice. The details are important not only throughout the design phases but must be carefully observed during build-out so that special isolation requirements are incorporated correctly. The entire process from design, detailing, and into construction is presented with emphasis on the various communication processes required to arrive at successful outcomes. Specific examples are provided highlighting the importance of each step in the process.

4:50

4pAAb4. Increasing liveness and clarity in a multipurpose civic center. Hans Michel and Joseph Myers (Acoust., Kirkegaard Assoc., 801 W. Adams 8th Fl., Chicago, IL 60607, hmichel@kirkegaard.com)

The Des Moines Civic center recently underwent a renovation primarily for symphonic uses, to improve hearing conditions on stage and in the house. This renovation included a new movable orchestra shell, rebuilt rear wall, and new finishes at sidewalls. The original shell did not adequately support bass or project sound to the audience, and musicians reported communication problems on stage. The rear wall was typically covered by a curtain, and returned confusing echoes when the curtain was withdrawn. The most unusual aspect of the room was the design of its sidewalls, broken into convex, down-leaning pylons, a signature of Paul Veneklasen, the original acoustician. KA analyzed the reflection pattern off the side walls and devised a treatment that maintained the beneficial aspects of the original design while dramatically improving clarity and presence. This paper focuses on the testing and modeling methods used to confirm our hypotheses about the problems caused by side wall reflections; the selection of appropriate materials based on in situ measurements and subjective listening; and architectural coordination.

Contributed Papers

5:10

4pAAb5. Acoustical considerations in shipping container architecture. Anat Grant (CSDA Design Group, 4061 Glencoe Ave., Ste. B, Marina del Rey, CA 90292, agrant@csdadesigngroup.com) and Randy Waldeck (CSDA Design Group, San Francisco, CA)

The inherent strength, wide availability, and relatively low expense of intermodal freight containers have led to an increase in their popularity as framing elements for modular construction. In school expansion and renovation projects, modular construction can reliably achieve sustainability and energy efficiency targets, often with shorter construction schedules. Residential housing, offices, commercial buildings and other building types are also constructed using shipping containers. While shipping container architecture is a green alternative with many advantages, it can also present acoustical challenges in buildings occupied by people. Low ceiling heights often limit acoustical treatment possibilities, and acoustical test data on sound isolation is limited. Case studies and measurement data presented will include an elementary school with large divisible classrooms, a community education center, an after-school youth education and classical music program in an economically struggling Los Angeles community, and a multifamily apartment building in Berkeley, CA.

5:25

4pAAb6. Silent Room: A (non)sensory interactive art installation. Matthew C. Zeh, Fiona Cheung (Mech. Eng. Dept., Univ. of Texas at Austin, 204 East Dean Keeton St., Austin, TX 78712-1591, mzeh@utexas.edu), Simon Heijdens (Artist/Creator of Silent Rm., Studio Simon Heijdens Ltd., London, United Kingdom), and Preston S. Wilson (Mech. Eng. Dept., Univ. of Texas at Austin, Austin, TX)

Simon Heijdens, a London-based Dutch artist, was commissioned by South by Southwest (SXSW), an annual film, media, and music festival in Austin, TX, to create an installation artwork to premiere at SXSW in March 2016. In reaction to the sensory overload of the festival, he created Silent Room. Students and professors from the University of Texas at Austin worked alongside Mr. Heijdens to design a portable, self-sufficient, reusable space providing sound isolation and absorption from outside and inside the chamber, respectively. Housing the space in a shipping container, Silent Room was successfully built and showcased at SXSW in downtown Austin to hundreds of visitors, achieving a minimum sound reduction of 27 dBA at 40 Hz and a maximum reduction of 55 dBA at 1.45 kHz. These results were achieved using a suspended floor and decoupled wall/ceiling construction of Auralex rubber U-Boats, 2" x 4" wood studs, mineral fiber insulation, plywood, Auralex SheetBlok vinyl, and acoustically treated gypsum board. After initial acoustical measurements, more U-Boats were added, decoupling the container underside from the street and isolating it from ground-borne vibration. However, the room performance worsened after this change, possibly due to exposure to a new air-borne path for noise ingress.

Session 4pABa

Animal Bioacoustics, Signal Processing in Acoustics, and Speech Communication: Sequence Information in Mammalian Vocal Call Production II

Marie A. Roch, Cochair
Dept. of Computer Science, San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720

Tadamichi Morisaka, Cochair
School of Marine Science and Technology, Tokai University, 3-20-1, Orido, Shimizu-ku, Shizuoka 4248610, Japan

Invited Papers

1:00

4pABa1. Fin whale call sequence analysis from tracked fin whales on the Southern California Offshore Range. Glenn Ierley (Scripps Inst. of Oceanogr., Houghton, MI) and Tyler A. Helble (SSC-PAC, 2622 Lincoln Ave., San Diego, CA 92104, tyler.helble@gmail.com)

Time difference of arrival (TDOA) methods were utilized to localize and track fin whales on the Southern California Offshore Range (SCORE). In several instances two whales were shown to converge at the same location on the range. Analysis of 20 Hz call sequences revealed changes in call type and intercall-interval coinciding with changes in swim kinematics of the nearby transiting whales. Call sequence analysis may provide some insight on the ability of fin whales to convey information to conspecifics on location and bearing of travel.

1:20

4pABa2. Blue whale song sequencing off Southern California. Leah A. Lewis (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0205, lalewis@ucsd.edu), John Calambokidis (Cascadia Res. Collective, Olympia, WA), John A. Hildebrand, and Ana Širović (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Blue whales off Southern California produce a unique song consisting of pulsed A unit and tonal B unit calls. These units are either sequenced in an alternating AB pattern, or a single A unit is followed by multiple B units (here denoted ABB). To investigate whether there is geographic variation in the occurrence of these different sequence types, data were analyzed from four High-frequency Acoustic Recording Packages deployed at two sites each inshore and offshore of the Channel Islands from September 2009 to June 2010. Additionally, fine-scale behavior associated with song and its sequencing was analyzed from acoustic tags deployed on blue whales in Southern California since 2000. A higher proportion of song bouts detected in offshore recordings were type ABB, whereas the dominant song type observed inshore was AB. This pattern may indicate geographic variability in song function. Most song units recorded on tags were produced during surface-travel dives, with the whale less than 30 m deep. Song bouts detected on tags occurred primarily at night, and whales did not interrupt their calling sequence with breathing intervals. The observed differences in calling and song preference off Southern California may be useful in identifying regionally distinct behavioral contexts for blue whales.

Contributed Papers

1:40

4pABa3. Central and western Pacific blue whale song and occurrence. Pollyanna I. Fisher-Pool, Erin Oleson (NOAA Fisheries, Pacific Islands Fisheries Sci. Ctr., 1845 Wasp Blvd., Bldg. 176, Honolulu, HI 96818, pollyanna.fisher-pool@noaa.gov), and Ana Širović (Scripps Inst. of Oceanogr., La Jolla, CA)

Blue whale (*Balenoptera musculus*) occurrence in the central and western Pacific is not fully understood. However, passive acoustics offer an effective way to monitor remote sites. Blue whale songs are regionally distinct, and their stereotyped characteristics may be used to distinguish populations. Most blue whale song consists of multiple, pulsed and tonal units. However, song in the central and western Pacific consists of only simple tonal units. We investigated song variability of Hawaii, Wake Atoll, Palmyra, and Tinian in the Mariana Islands Archipelago, and highlight differences between blue whale songs amongst sites. Song calls of two different frequencies were recorded by High-frequency Acoustic Recording Packages. Detailed measures of call frequency were taken along the contours of tonal calls and patterning of song sequences was evaluated for recordings from the four locations during 2012 and 2013. The spatial and temporal occurrence of these patterns is discussed.

1:55

4pABa4. Stereotyped, repetitive gunshot call patterns produced by the North Pacific right whale, *Eubalaena japonica*. Jessica Crance and Catherine Berchok (AFSC/NMFS/NOAA, National Marine Mammal Lab., 7600 Sand Point Way NE, Seattle, WA 98115, Jessica.Crance@noaa.gov)

The Marine Mammal Laboratory has deployed long-term passive acoustic recorders along the 50 m and 70 m isobaths throughout the Bering Sea since 2007. Instruments recorded at either 4 kHz on a ~7-11% duty cycle, or 8 kHz on a 30-45% duty cycle. In addition, directional sonobuoys were

4p THU. PM

deployed during field surveys to allow real-time monitoring for large whale presence. During the 2010 survey, a stereotyped, repetitive gunshot call pattern was acoustically detected on sonobuoys. This same call pattern was then detected on two different long-term moored recorders in two separate years. Since then, six different stereotyped, repetitive patterns have been documented, three of which have been analyzed. Preliminary results show that each pattern has a minimum of 30 iterations repeated over several hours. Furthermore, in several cases, these patterns are repeated throughout the season, in consecutive years, and in one instance, in non-consecutive years. While male North Atlantic right whales produce long gunshot bouts similar to the reproductive advertisement known in other species, right whales are not known to produce any type of repeated stereotyped pattern. This represents the first study to document stereotyped repetitive gunshot patterning in right whales. [Work funded by the Bureau of Ocean Energy Management.]

2:10

4pABa5. Fin whale song patterns in Southern California and the Gulf of California. Ana Sirovic (Scripps Inst. of Oceanogr., 9500 Gilman Dr. MC 0205, La Jolla, CA 92093-0205, asirovic@ucsd.edu), Erin M. Oleson (Pacific Islands Fisheries Sci. Ctr., Honolulu, HI), Jasmine S. Buccowich, and Ally Rice (Scripps Inst. of Oceanogr., San Diego, CA)

The most common fin whale calls, 20 Hz pulses, are often produced in regular, stereotypic sequences termed songs. The main variability in songs comes from the differences in the duration of the interval between successive pulses, the interpulse interval (IPI). Data recorded between 2000 and 2012 in Southern California and 2004 to 2010 in the Gulf of California were analyzed for temporal and geographic patterns in fin whale songs. During this time, fin whales in Southern California and the Gulf of California produced four song types with distinct IPI sequences. Two common songs in Southern California were the short doublet and the long doublet and they were detected year-round. The IPIs in the short doublet song have been increasing over the long term, while the IPIs in the long doublet song showed no long-term trends but were seasonally variable. In the Gulf of California, the most common songs were the short triplet and the long triplet. These songs were detected in Gulf of California year-round, although their occurrence decreased August through October. There was some exchange of sons between the two areas that might point to seasonal movement of parts of these populations.

2:25

4pABa6. Fin whale occurrence and population structure in the central and western Pacific through detection and characterization of song patterns. Erin M. Oleson (Protected Species Div., NOAA Fisheries, Pacific Islands Fisheries Sci. Ctr., 1845 Wasp Blvd. B176, Honolulu, HI 96818, erin.oleson@noaa.gov), Ana Širović, Ally Rice, and Leah M. Varga (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Fin whale songs comprise series of 20Hz pulses and song structure can be characterized by patterns of inter-pulse intervals (IPIs). Recent studies in the northeast Pacific indicate the occurrence of a broadly distributed, seasonally changing, but annually stable IPI pattern, heard from the Bering Sea to southern California and west to Hawaii. Other geographically localized song patterns are also known in the Gulf of California, the Bering Sea, and off southern California. We expanded the assessment of fin whale song across the Pacific, analyzing occurrence and song patterns from multi-year passive acoustic data collected at several sites in the central and western Pacific, from Palmyra Atoll to the Northwestern Hawaiian Islands and west including Wake Atoll and the Mariana Archipelago. Fin whales are relatively uncommon in these tropical and sub-tropical waters, though they are detected in all seasons at some sites. Song IPI patterns suggest multiple and potentially annually variable fin whale song structures throughout the region. The common northeast Pacific song is heard at Hawaii and Wake Atoll sites, with different song types at Palmyra and in the Mariana Archipelago. Long-term temporal and geographic patterns of song occurrence across the region will be discussed.

Session 4pABb

Animal Bioacoustics: Session in Honor of Whitlow Au III

Kerri Seger, Chair
Scripps Institution of Oceanography, 9331 Discovery Way, Apt. C, La Jolla, CA 92037

Contributed Papers

2:55

4pABb1. Comparison of three *Stenella* spp. audiograms measured using auditory evoked potentials and behavioral methods. Danielle R. Greenhow (School of Ocean Sci. and Technol., Univ. of Southern MS, 2012 W 2nd St., Apt. 320, Long Beach, MS 39560, danielle.greenhow@usm.edu), Adrienne Cardwell (Mote Marine Lab., Clearwater, FL), Jessica R. Powell (Mote Marine Lab., Saint Petersburg, FL), Mandy L. Hill-Cook (Dept. of Biology, Portland State Univ., Portland, OR), M. Andrew Stamper (Mote Marine Lab., Orlando, FL), Lynne Byrd (Mote Marine Lab., Sarasota, FL), Charles A. Manire (Mote Marine Lab., Juno Beach, FL), Micah C. Brodsky (Micah Brodsky, V.M.D. Consulting, Miami Shores, FL), Gordon B. Bauer (Div. of Social Sci., New College of Florida, Sarasota, FL), and David A. Mann (College of Marine Sci., Univ. of South Florida, Sarasota, FL)

The oceanic dolphin genus *Stenella* is underrepresented in cetacean hearing data. In this study the hearing of three *Stenella* spp. dolphins was measured using auditory evoked potential (AEP) methods. A single male juvenile Atlantic spotted dolphin (*Stenella frontalis*) was rehabilitated in Key Largo, Florida, after being sighted alone and emaciated. The Atlantic spotted dolphin's greatest sensitivity was at 40 kHz, with functional hearing up to 128 kHz. A female spinner dolphin (*Stenella longirostris*), housed at Mote Marine Laboratory, had peak sensitivity at 40 kHz and functional hearing up to 120 kHz, the highest frequency tested. The sensitive high frequency hearing of the spinner and Atlantic spotted dolphin is similar to that found in other oceanic odontocetes. A pantropical spotted dolphin (*Stenella attenuata*) was housed at Mote Marine Laboratory and hearing thresholds were determined using AEP and behavioral methods. The pantropical spotted dolphin had a peak sensitivity at 10 kHz, with a cutoff frequency between 14 and 20 kHz. The source of the dramatic high-frequency hearing loss is not known; possible causes include congenital hearing loss (present at birth) or hair cell death due to treatment with ototoxic drugs during rehabilitation.

3:10

4pABb2. Functional optical imaging from bat inferior colliculus using a micro-endoscope. Hidetaka Yashiro (Doshisha Univ., 1-3, Miyakodani, Tatara, Kyotanabe, Kyoto 6100321, Japan, hidetaka.yashiro@gmail.com), Kazuo Funabiki (Inst. of Biomedical Res. and Innovation, Kobe, Japan), Andrea M. Simmons, James A. Simmons (Brown Univ., Providence, RI), and Hiroshi Riquimaroux (Shandong Univ., Tokyo, Japan)

We developed a confocal micro-endoscopic system that enables simultaneous recording of electrical neuronal responses—local field potentials (LFPs) and multi-unit activities, (MUA) — and fluorescence changes reflecting calcium activation derived from Oregon green dye. We recorded these responses from different depths of the inferior colliculus (IC) in two species of bats (*Carollia perspicillata* and *Eptesicus fuscus*) to tone bursts, noise bursts, and FM sweeps. Electrical responses were fast and registered the occurrence of sounds with submillisecond and millisecond precision. LFPs contained strong onset responses and weaker, extended oscillations that persisted for much of the sound's duration. In contrast, calcium responses were slow, building up gradually over the course of 50-100 ms, and declined slowly following sound offset. Full horizontal-vertical video image scanning of the entire fiber optic bundle's face was limited to 18.9 frames/s by the scanning mirror, but isolation and recording of successive individual horizontal scans, done by freezing the vertical scan, yielded 7,550 lines/s, or 132 microseconds for 1 line. Line-scanning revealed that the calcium responses indeed were slow, in the 50-100 ms range. They also were spatially diffuse, suggesting a distributed dendritic origin. [Work supported by JSPS, MEXT Japan, Shandong University, ONR, Capita Foundation.]

3:25

4pABb3. Human listening experiments provide insight into cetacean auditory perception. Caroline M. DeLong (Psych., Rochester Inst. of Tech., 18 Lomb Memorial Dr., Rochester, NY 14623, cmdgsh@rit.edu)

Odontocete cetaceans produce an array of sounds including echolocation clicks that can be used for object recognition and whistles used primarily for communication. Echoes and whistles contain a multitude of acoustic features, and the salient features used by animals when identifying objects or in social contexts are often difficult to isolate. One method for detecting salient acoustic features is to compare the performance of cetaceans and humans on the same auditory perception tasks. Human listeners can be presented with echoes produced with simulated cetacean clicks or whistle-like stimuli and provide verbal feedback on discriminatory cues. An early human listening study was performed by Au and Martin (1989). Seven studies performed over the past 15 years show that humans perform as well or better than cetacean subjects in a variety of tasks: echoic discrimination of objects varying in size, shape, material, texture, or wall thickness; echoic recognition of objects from various aspect angles; echoic object discrimination using clicks from different cetacean species; and recognition of frequency-modulated whistle-like sounds. Analyzing the error patterns of humans and cetaceans alongside the reported cues reveals processing mechanisms and decision strategies cetaceans may use. This comparative approach sheds light on how cetaceans perceive and represent sounds.

3:40

4pABb4. Acoustic processes in an echolocating bottlenose dolphin's (*Tursiops aduncus*) head using a finite element model. Chong Wei (College of Ocean and Earth Sci., Hawaii Inst. of Marine Biology, Xiamen Univ., Lilipuna Rd., Xiamen, Fujian 361102, China, weichong3310@foxmail.com), Whitlow W. Au (Hawaii Inst. of Marine Biology, Kaneohe, HI), Zhongchang Song (College of Ocean and Earth Sci., Xiamen Univ., Xiamen, Fujian, China), Darlene R. Ketten (Woods Hole Oceanographic Inst., Woods Hole, MA), and Yu Zhang (College of Ocean and Earth Sci., Xiamen Univ., Xiamen, Fujian, China)

Bottlenose dolphins (*Tursiops aduncus*) are a well-known species using broadband echolocation signals for searching prey and spatial orientation. In this study, the computed tomography (CT) scan data were obtained to set up a two-dimensional finite element model. In the vertical plane, the acoustic field on the animal's forehead and the far field transmission beam pattern of an echolocating dolphin were calculated. The simulation results and prior measurement results were consistent qualitatively. The role of the main structures on the sound propagation pathway such as air sacs, melon, skull,

and connective tissues was investigated. Furthermore, the signal at the source excitation was investigated. It suggested that the broadband echolocation dolphins may not have the same driving signals at the source excitation as the narrowband echolocation dolphins. The results can help us gain further understanding of the acoustic processes in dolphin's biosonar.

3:55

4pABb5. Ultra-high-frequency hearing in seals and sea lions in the context of other secondarily aquatic mammals. Colleen Reichmuth (Inst. of Marine Sci., Long Marine Lab., Univ. of California Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060, coll@ucsc.edu), Jason Mulsow (National Marine Mammal Foundation, San Diego, CA), Jillian M. Sills, Brandon L. Southall, and Kane Cunningham (Inst. of Marine Sci., Long Marine Lab., Univ. of California, Santa Cruz, CA)

The underwater hearing abilities of marine mammals near the upper frequency limit of hearing are poorly understood. While the maximum frequency of hearing is often identified following a sharp roll-off in sensitivity, it is apparent that many marine mammals can perceive high-amplitude tonal sounds above this frequency region. We recently measured underwater detection thresholds in quiet conditions in the 50-180 kHz range for one California sea lion, one spotted seal, and one harbor seal. Absolute sensitivity curves from these subjects exhibited two distinct slope regions at frequencies near the high-frequency hearing limit. The first region is characterized by an initial, rapid decrease in sensitivity with increasing frequency—*i.e.* a steep slope—while the second shows a much less rapid sensitivity decrease—*i.e.* a shallower slope. An additional masking study conducted with one of the seals suggested that the initial, rapid decrease in sensitivity is not due to cochlear constraints, as has been proposed, but rather due to constraints on the underlying bone-conduction mechanism. These results can be viewed in the context of the selective pressures influencing ultra-high-frequency hearing in other aquatic mammals, including those capable of specialized echolocation.

4:10

4pABb6. Killer whale (*Orcinus orca*) audiograms. Brian K. Branstetter (National Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106, brian.branstetter@nmmpfoundation.org), Doug Acton (Sea World San Antonio, 10500 Sea World Dr., San Antonio, CA), John Stewart (Sea World San Diego, 500 Sea World Dr., San Diego, CA), Dorian Houser (National Marine Mammal Foundation, San Diego, CA), Judy St. Ledger (Sea World San Diego, 500 Sea World Dr., San Diego, CA), James Finneran, and Keith Jenkins (US Navy Marine Mammal Program, SSC Pacific, Code 17519, 53560 Hull St., San Diego, CA)

Historically, understanding of killer whale (*Orcinus orca*) hearing was based on behavioral and evoked potential data from three animals, one of which had significant hearing loss. For the other two whales, the mean detection threshold at 20 kHz was 36 dB re 1 μPa; this is the lowest underwater behavioral detection threshold of any marine mammal tested, suggesting that it may be an outlier. The current study measured the behavioral audiograms of eight killer whales at two different facilities. Hearing sensitivity was measured from 100 Hz to 160 kHz in animals ranging in age from 12 to 52 years. Two whales had hearing loss consistent with presbycusis and a third displayed atypical low-frequency hearing loss (but normal high frequency hearing); previously measured low thresholds at 20 kHz were not replicated in any animal. Hearing in the killer whales was generally similar to other delphinids, with lowest threshold (49 dB re 1 μPa) at approximately 40 kHz, good hearing (i.e., within 20 dB of best sensitivity) from 5—80 kHz, and low- and high-frequency hearing cut offs ($>$ 100 dB re μPa) of 600 Hz and 114 kHz, respectively.

4:25

4pABb7. Control ultrasound beam by tissues in the head of finless porpoise acting as a tunable gradient index material. Yu Zhang, Zhongchang Song, Chong Wei (Dept. of Appl. Phys. and Eng., College of Ocean and Earth Sci., Xiamen Univ.,Rm. C3-209, Xiping Bldg., Xiangan Campus of Xiamen Univ., Xiangan South Rd., Xiangan District, Xiamen, Fujian 361102, China, yuzhang@xmu.edu.cn), Whitlow W. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kaneohe, HI), Wenwu Cao (Dept. of Mathematics and Mater. Res. Inst., The Penn State Univ., Pennsylvania, PA), and Xianyan Wang (Lab. of Marine Biology and Ecology, Third Inst. of Oceanogr., State Oceanic Administration, Xiamen, Fujian, China)

Porpoises are well known to emit directional ultrasound beams for detecting and tracking preys; however, how they produce and manipulate directional beams are challenging. Here, we investigated physical mechanism of ultrasound beam formation and control of finless porpoise (N. a. sunameri) by using an integrated scheme of computed tomography, tissue and field measurements, and numerical modeling. The results showed that complex acoustic structures in the porpoise's forehead contributed to producing directional acoustic field. Furthermore, we demonstrated that the skull, air sacs, connective tissue, muscle, and melon constituted a gradient index (GRIN) structure whose density and sound velocity are positively correlated, and thus regulated the directional beam. The removal or compression deformation of the forehead tissues decentralizes energy and widens sound beam, indicating that the forehead tissues as a tunable natural GRIN material significantly impact beam patterns of the finless porpoise. The results might be valuable for understanding control mechanism of acoustic beam of other toothed whales.

4:40

4pABb8. Multi-species Baleen Whale audiogram modelling. Darlene R. Ketten (Otology and Laryngology, Harvard Med. School, Woods Hole Oceanographic Inst., Woods Hole, MA 02543, dketten@whoi.edu), Aleks Zosuls, and Andrew Tubelli (Biomedical Eng., Boston Univ., Boston, MA)

In this research, we produced model audiograms for two Mysticetes (baleen whales) that are among the species most likely to be subject to impacts from common lower frequency anthropogenic sound sources deployed in the oceans. These models are needed for species-specific risk assessments for hearing impacts, for determining optimal signals for playback experiments, and for determining effective electrode and sound source placements for auditory brainstem response (ABR) measures in live stranded whales. We employed micro and UHRCT, dissection, and histology of minke (*Balaenoptera acutorostrata*) and humpback (*Megaptera novaeangliae*) heads and ears to calculate inner ear frequency maps for determining total hearing range, the frequency of peak sensitivity, and the probable frequency of NIHL liability ("notch"). The anatomically derived data were then combined with direct measures via nanoindentation of middle ear stiffness, Young's modulus, frequency response, and inner ear stiffness to determine the middle ear transfer function and basilar membrane stiffness gradients. FEM simulations were employed to further explore the responsivity of the tympanum and middle ear ossicular chain. In order to test the validity of the model techniques, model audiograms were produced also for two odontocetes, the bottlenose dolphin and the harbor porpoise, and compared with well-established behaviorally obtained audiograms for these species. [This research was supported by the LMR and JIP marine research programs.]

4:55

4pABb9. Acoustic properties reconstruction of forehead tissues in an Indo-Pacific humpback dolphin (Sousa chinensis), with investigation on temperature effects on the species tissues sound velocity and beam. Zhongchang Song, Yu Zhang, Chong Wei (Dept. of Appl. Phys. and Eng., College of Ocean and Earth Sci., Xiamen Univ.,Rm. C3-210, Xiping Bldg., Xiangan Campus of Xiamen Univ., Xiangan South Rd., Xiangan District, Xiamen, Fujian 361102, China, songzhongchang@foxmail.com), and Per Berggren (Dove Marine Lab. and School of Marine Sci. and Technol., Newcastle Univ., Newcastle, United Kingdom)

Computed tomography (CT) imaging and ultrasound experimental measurements were used to reconstruct the acoustic properties (density,

velocity, and impedance) of forehead tissues from a deceased Indo-Pacific humpback dolphin (Sousa chinensis). The nonlinear regression methods were used to demonstrate the relationships between the sound velocity and temperature in melon, muscle and connective tissue. The obtained nonlinear equations were then combined with the original CT scanning results and sound velocity distributions reconstructed at room temperature 25°C to reconstruct the dolphin head's sound velocity distribution at temperature 37°C. The beam formation and beam properties between two temperatures 37°C and 25°C were then compared and discussed. The results could provide important information for understanding the species' bioacoustic characteristics and the acoustic data can be used for investigation of biosonar beam formation of this species.

5:10

4pABb10. Say what? Hearing abilities and subsequent variability for a wild population of Beluga Whales (*Delphinapterus leucas*). T. Aran Mooney (Biology Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, amooney@whoi.edu), Manuel Castellote (Alaska Fisheries Sci. Ctr., National Marine Mammal Lab., NMFS, Seattle, WA), Lori Quakenbush (Alaska Dept. of Fish and Game, Fairbanks, AK), Roderick Hobbs (Alaska Fisheries Sci. Ctr., National Marine Mammal Lab., NMFS, Seattle, WA), Caroline Goertz (Alaska SeaLife Ctr., Seward, AK), and Eric Gaglione (Georgia Aquarium, Atlanta, Georgia)

Odontocetes use sound for vital biological functions and hearing is considered their primary sensory modality. To understand and predict noise impacts, we need to know natural hearing ability at the population-level. Yet we have no wild population-level hearing data for any cetacean. Here, we measured the audiograms (4-150 kHz) of wild beluga whales from 2012-2016 using auditory evoked potential methods. The goal of this work was to establish the mean audiogram and variance of wild beluga whales to define what this population hears. Audiograms were collected from 26 belugas during capture-release events in Bristol Bay, AK. In general, belugas showed excellent sensitivity (<60 dB) from 45 to 100 kHz and most (76%) heard up to at least 120 kHz. Hearing thresholds at lower frequencies 4-11 kHz tracked the low environmental noise levels (70 dB/Hz), suggesting that quiet habitats may allow substantial hearing sensitivity. Beluga audiograms show substantial (20-40 dB) variation but less than that of bottlenose dolphins suggesting possible differences between populations and the need for comparative data. Further, the hearing variability among belugas suggests multiple measures best describe the maximum sensitivity and population variance. These audiograms quadruple the data for beluga whales as a species and provide the beginning of population-level auditory data for a wild odontocete.

THURSDAY AFTERNOON, 1 DECEMBER 2016 — CORAL 3, 3:00 P.M. TO 5:30 P.M.

Session 4pABc

Animal Bioacoustics: Animal Bioacoustics Poster Session

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

All posters will be on display from 3:00 p.m. to 5:30 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 3:00 p.m. to 4:15 p.m. and authors of even-numbered papers will be at their posters from 4:15 p.m. to 5:30 p.m.

Contributed Papers

4pABc1. Ultrasonic vocalizations in laboratory mice. Kali Burke, Laurel A. Screven, and Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, 246 Park Hall, Buffalo, NY 14260, kaliburk@buffalo.edu)

Mice produce diverse ultrasonic vocalizations (USVs), which vary in spectrotemporal parameters including frequency, intensity, and duration. Although it is currently unclear whether mice are using these USVs for communication, researchers have attempted to parse USVs into categories based upon their spectrotemporal characteristics. There is recent evidence that shows that mice are producing context-specific vocalizations, but it is still unclear if USVs have any of the language-like characteristics that researchers would like to impose on them. The current literature clearly indicates a need for a comprehensive catalog of the USVs mice are producing in order to better understand if and how these animals are using their USVs for communication. Here, 20 male and 20 female mice were exposed to a same sex or opposite sex stimulus mouse for 1 hour and recorded immediately following separation for 3 min. All vocalizations were categorized based on their spectrotemporal characteristics. This experiment provides a catalog of the vocal repertoire of laboratory mice, adding to the existing knowledge of how they are using their USVs.

4pABc2. Discrimination of natural from synthetic ultrasonic vocalizations by laboratory mice. Anastasiya Kobrina, Laurel Screven, and Micheal Dent (Psych., SUNY Univ. at Buffalo, B23 Park Hall, Amherst, NY 14261, akobrina@buffalo.edu)

Mice produce spectrotemporally complex ultrasonic vocalizations (USVs) which are thought to be important for mate attraction, courtship, and other social interactions. Despite this assertion, little is known about how mice perceive or use these auditory signals. USVs have been categorized based on a variety of spectrotemporal parameters, including frequency, amplitude, and duration. Previous studies have established that mice are capable of detecting and discriminating natural, synthetic, and altered USVs using physiological and behavioral methodologies. The current research examined whether mice are capable of discriminating natural USVs from their synthetic analogs. Discrimination performance was tested in six adult mice using operant conditioning procedures with positive reinforcement. Discrimination was high for some USV categories but low for others, suggesting that more than simple spectrotemporal similarity is involved in the discrimination of USVs.

4pABc3. Acoustic multi-tasking in beaked whales: The use of alternating echolocation regimes during the descent phase of deep foraging dives. Jennifer S. Trickey, Simone Baumann-Pickering, and John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, jtrickey@ucsd.edu)

Beaked whales use echolocation to navigate and hunt at extreme depths, and the timing of their echolocation clicks while actively searching for prey is generally characterized by a stable inter-click interval (ICI). However, analysis of autonomous, long-term passive acoustic data revealed a markedly different pattern during the vocal portion of their descent towards the seafloor. Diving beaked whales were found to alternate between two ICI regimes, while also gradually increasing their overall click rate. This strategy is presumably used to simultaneously monitor two different target ranges as they approach their preferred foraging depth. One ICI regime likely corresponds to the two-way-travel time of sound to the seafloor, and by examining the rate at which the time interval between clicks decreased in this seafloor-tracking ICI regime, we calculated estimates of dive descent rate. The second ICI regime was consistently more rapid, and represents a shorter search range that is likely used by the whale to inspect the nearby water column for the presence of prey and other features. This unique echolocation behavior was identified in acoustic encounters of four different species of beaked whales recorded at a variety of sites in the North Pacific Ocean and the Gulf of Mexico.

4pABc4. Detection of marine mammal vocalizations off Cape Hatteras, modulated by the Gulf Stream. Stephen B. Lockhart (Marine Sci., Univ. of North Carolina, CB#3300, Chapel Hill, NC 27599-3300, sblockhart.z0@gmail.com), Lindsay L. Dubbs, Michael Muglia, Patterson Taylor (Univ. of North Carolina Coastal Studies Inst., Wanchese, NC), and Robert E. Todd (Woods Hole Oceanographic Inst., Woods Hole, MA)

To assess the ecological impact of extracting energy from the Gulf Stream, the University of North Carolina Coastal Studies Institute has deployed a mooring on the continental slope off Cape Hatteras at a depth of 230 m, equipped with an ADCP, a CTD, and a hydrophone. Analyzing data from the first deployment, we automatically detected marine mammal vocalizations. We found a strong correlation between the amount of vocalizations and the position of the Gulf Stream, with vocalizations increasing when the Gulf Stream was farther offshore. We then derived sound speed profiles for the region around the mooring using temperature and salinity profiles collected by WHOI Spray gliders. These sound speed profiles were fed into a propagation modeling tool. When the Gulf Stream was far offshore with respect to the mooring position, the model showed a set of rays that propagate down the slope in an acoustic channel along the bottom. When the Gulf Stream was nearly over the mooring, the range of angles that could propagate down the slope was reduced by a factor of six. Therefore, the propagation models suggest that the observed correlation is due to a propagation effect, resulting from the varying position of the Gulf Stream.

4pABc5. Emergence density and stream frequency spectra of *Tadarida brasiliensis*. Stephanie Dreessen, Cassi Mardis, and Laura Kloepper (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, sdrees01@saintmarys.edu)

Bats depend on echolocation to maneuver through their environment. Mexican Free-Tailed bats (*Tadarida brasiliensis*) form large maternal colony roosts and emerge in large, dense groups. Previous work found that the *T. brasilensis* bats adjust their call structure during emergence, which may help the bats avoid flight collisions. The limited physical space between bats in the emerging stream is not only problematic with flight collisions, but also with echolocation interference. In this study, we tested how the density of emergence affects echolocation. Specifically, we tested the hypothesis that the density of flying bats affects echolocation characteristics of the entire stream. We determined six emergence density categories from thermal imagery, and randomly selected 20, 500 ms audio samples corresponding to each density. For each audio sample, we calculated the frequency spectrum, and analyzed how the stream frequency spectrum changed according to population density. Across densities, the stream frequency spectra all have the same shape and peak frequencies, but the variance for certain frequencies differs. This change in variance suggests that these bats are attempting to avoid jamming in dense emerging streams by altering their frequencies from bat-to-bat, irrespective of density.

4pABc6. A first look at the Ocean Noise Reference Station Network: The soundscape of Stellwagen Bank National Marine Sanctuary. Samara M. Haver (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., 2030 SE Marine Sci. Dr., Newport, OR 97365, samara.haver@oregonstate.edu), Holger Klinck (BioAcoust. Res. Program, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY), Sofie Van Parijs (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA), Haruyoshi Matsumato, Robert P. Dziak (NOAA Pacific Marine Environ. Lab., Newport, OR), Jason Gedamke (Office of Sci. and Technol., NOAA Fisheries, Silver Spring, MD), and Leila T. Hatch (Gerry E. Studds Stellwagen Bank National Marine Sanctuary, NOAA Office of National Marine Sanctuaries, Scituate, MA)

In order to compare long-term changes and trends in soundscapes around the United States, a multi-year network of identical autonomous passive acoustic recording systems, the Ocean Noise Reference Station (NRS) Network, has been established. In partnership with the National Oceanic and Atmospheric Administration (NOAA) Office of Oceanic and Atmospheric Research, NOAA Pacific Marine Environmental Laboratory, NOAA National Marine Fisheries Service, NOAA Office of National Marine Sanctuaries, and the National Park Service, hydrophone moorings were deployed in 12 discrete soundscapes in the Northeast Pacific and Northwest Atlantic oceans to record underwater ambient sound levels in the 10 to 2200 Hz frequency range. This initial analysis utilized the first year of available data from the NRS deployed in Stellwagen Bank National Marine Sanctuary (SBNMS), a busy area for both natural and anthropogenic activity. Preliminary results indicate that: (1) broadband (10-2200 Hz) ambient sound levels at SBNMS are stable year round in association with constant vessel activity in the nearby shipping lane to Boston, MA, (2) surface wind speed is positively correlated with broadband noise levels, (3) intensity of low-frequency baleen whale calling activity varies seasonally, but signals are acoustically detectable year-round. Future analyses will compare the soundscapes of all 12 NRS sites.

4pABc7. Analysis of fin whale vocalizations south of Rhode Island. Jennifer L. Giard, Makio Tazawa, James H. Miller, Gopu R. Potty (Ocean Eng. Dept., Univ. of Rhode Island, 215 S Ferry Rd., Narragansett, RI 02882, jennifer_giard@my.uri.edu), Ying-Tsong Lin, Arthur Newhall, and Mark Baumgartner (Woods Hole Oceanographic Inst., Woods Hole, MA)

Fin whale vocalizations were recorded south of Rhode Island during late summer through early fall of 2015 using a number of underwater recording systems. These systems were deployed to monitor broadband noise, including pile driving, from construction of the Block Island Wind Farm. Two vertical hydrophone array moorings were deployed in approximately 40 meters of water each with four hydrophones. Additionally, a tetrahedral array was deployed in about 30 meters of water just above the seabed. The tetrahedral array consisted of four hydrophones spaced 0.5 meters apart. The spacing between each of these recording systems was approximately 7.5 km. 20-Hz fin whale vocalizations were recorded numerous times on all of the sensors both during and after construction was completed. An analysis and localization effort of these signals was performed to estimate the source level, directionality, and the track of the whale over the period of the vocalizations. The results of this analysis will be discussed. [Work supported by the BOEM.]

4pABc8. Detection and localization of whale vocalizations in archived data of seafloor cabled observatories in eastern Japan. Ryoichi Iwase (CEAT, JAMSTEC, 3173-25 Showa-machi, Kanazawa-ku, Yokohama, Kanagawa 236-0001, Japan, iwaser@jamstec.go.jp)

Through the examination of archived OBS (Ocean Bottom Seismometer) waveform data of seafloor cabled observatories in the eastern Japan, fin whale vocalizations were detected. Most of those vocalizations were observed with a single OBS because of the sparseness of the OBS deployment. At the observatory off Kushiro-Tokachi in Hokkaido, the fin whale vocalizations were localized using waveform data of both OBS and hydrophone at the same site (Iwase, 2015). At the observatory off Kamaishi in Tohoku District composed of OBSs without a hydrophone, seismic wave velocities in sediment, which are necessary parameters in estimating

horizontal range of sound source from apparent emergence angle of transmitted wave at seafloor, were estimated *in situ* based on air gun signal observation (Iwase, 2016). By using those results, the fin whale vocalizations were localized. At the observatory off Hatsushima Island in Sagami Bay, sperm whale vocalizations were detected in the audible sound signals of a single hydrophone which were recorded on the soundtrack of video tapes with seafloor images. To date, more than 7000 video tapes recorded since 1993 have been archived, and the result of analysis in progress will be shown at the presentation. References: Iwase (2015), DOI: 10.7567/JJAP.54.07HG03. Iwase (2016), DOI: 10.7567/JJAP.55.07KG01.

4pABc9. Improvements to using inter-click intervals to separate odontocete click trains from multiple animals. Jeremy Young, Anders Høst-Madsen (Dept. of Elec. Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 483, Honolulu, HI 96822, jbyoung@hawaii.edu), and Eva-Marie Nosal (Dept. of Ocean & Resources Eng., Univ. of Hawaii at Manoa, Honolulu, HI)

At the December 2013 ASA meeting, we introduced a method to separate click trains from multiple odontocetes that relies on click timing only. The method assumes that ICIs are slowly-varying according to some distribution. Click trains are classified by sequentially going through the clicks and maximizing the likelihood based on these distributions. Since then, additions have been made to improve performance and make it more practical including a source number estimation method, a refinement to the algorithm when the ICI distributions are unknown, and a way to incorporate information from the shape of clicks. We will present the results of these new methods on real and simulated datasets.

4pABc10. Investigating the occurrence of spinner dolphins around Maui Nui, Hawaii. Megan M. McElligott (Marine Biology Graduate Program, Univ. of Hawaii at Manoa, Hawaii Inst. of Marine Biology, 46-007 Lilipuna Rd., Kaneohe, HI 96744, meganmce@hawaii.edu), Marc O. Lammers (Hawaii Inst. of Marine Biology, Univ. of Hawaii at Manoa, Honolulu, HI), and Pollyanna Fisher-Pool (Cetacean Res. Program, NOAA Pacific Islands Fisheries Sci. Ctr., Ford Island, HI)

Spinner dolphins (*Stenella longirostris*) occur commonly along Hawaii's sloping coastlines where they follow a daily behavioral routine of foraging offshore at night on vertically migrating mesopelagic and epipelagic prey and then return to shallower waters to rest during the day. Populations of spinner dolphins have been well documented off Hawaii island and Oahu, but much less so off the other islands. In the Maui Nui region (Maui, Molokai, Lanai, and Kahoolawe), spinner dolphins are sighted regularly, but little is known about their use of the nearshore habitat, which is characterized by much shallower bathymetry than Hawaii island and Oahu. As a result, spinner dolphins in Maui Nui offer a unique opportunity to examine the adaptability of the species to a habitat that would not be considered ideal for their life history needs. To better understand the habitat-use patterns of spinner dolphins in Maui Nui, bottom moored acoustic recorders are being used to examine spinner dolphin presence off west Maui and southeast Lanai. These data are compared to similar data previously obtained from a well-studied spinner dolphin population off west Oahu. Preliminary results from these analyses are presented in combination with data on habitat characteristics and anthropogenic impact.

4pABc11. Sound localization of fish calls recorded by two stereo-underwater-recorders. Kazuki Yamato, Ikuo Matsuo (Tohoku Gakuin Univ., 2-1-1 Tenjinzawa, Izumi-ku, Sendai 981-3193, Japan, kyamato@mail.tohoku-gakuin.ac.jp), Ryuzo Takahashi (Japan Fisheries Res. and Education Agency, Kamisu, Japan), Naoto Matsubara, and Hiroki Yasuma (Hokkaido Univ., Hakodate, Japan)

Some fish produce sounds during the spawning and ramping behaviors. These sounds have been used for discrimination of fish species. Recently, the method was proposed to estimate the direction of fish call by using the arrival time difference between two hydrophones. However, it was difficult to estimate sound localization in the case of using the measured data from two hydrophones, that is, one stereo recorder. In this presentation, we used two stereo recorders, including four hydrophones for sound localization. The sound data were measured during spawning at fish, *Hexagrammos otakii*, in the littoral region of Usujiri-cho, Hakodate, Hokkaido, Japan (N41°56'11, E140°56'41). First, the fish calls were automatically detected by using the sound features including duration, max frequency, and amplitude. These features were determined by analyzing the sound data manually. Second, the arrival directions of detected fish calls were estimated in terms of each stereo recorder. Then, the fish calls were localized by using two arrival directions. As a result, it was clear that there were fishes producing call sounds around the spawning bed.

4pABc12. Source levels and calling rates for humpback whale (*Megaptera novaeangliae*) non-song vocalizations in Glacier Bay Alaska. Michelle Fournet (Fisheries and Wildlife, Oregon State Univ.- Hatfield Marine Sci. Ctr., 425 SE Bridgeway Ave., Corvallis, OR 97333, mbellalady@gmail.com), Holger Klinck (BioAcoust. Res. Program, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY), and Christine Gabriele (Humpback Whale Monitoring Program, Glacier Bay National Park, Gustavus, AK)

Marine resource managers in Glacier Bay National Park, Alaska, have been tasked with assessing the impact of vessel noise on marine mammal species. Humpback whales (*Megaptera novaeangliae*) are a highly vocal baleen whale that forage between spring and fall in Park waters. While calling rates and source levels for this species have been described on breeding grounds and migratory corridors, these fundamental acoustic parameters have not been thoroughly described on foraging grounds, and have never been empirically measured in the North Pacific. Using a three-element hydrophone array deployed in the Beardslee Island complex of Glacier Bay during the 2015 summer foraging season humpback whale "whup" and "growl" calls were localized and source levels were calculated. Array recordings were paired with shore based abundance estimates from the same period and calling rates (calls per whale per hour) were estimated. Known calling rates and source levels are essential for developing detection algorithms, masking metrics, and determining communication space on this historically important foraging ground.

4pABc13. Source levels of harbor seal underwater reproductive advertisement displays. Leanna P. Matthews (Biology Dept., Syracuse Univ., 114 Life Sci. Complex, 107 College Pl., Syracuse, NY 13244, lematthe@syr.edu), Jamie Womble (Glacier Bay National Park and Preserve, Juneau, AK), Christine Gabriele (Glacier Bay National Park and Preserve, Gustavus, AK), Holger Klinck (BioAcoust. Res. Program, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY), and Susan Parks (Biology Dept., Syracuse Univ., Syracuse, NY)

Harbor seals (*Phoca vitulina*), along with the majority of other phocid species, mate underwater. During the breeding season, male harbor seals set up underwater territories and use acoustic cues, known as roars, to defend these areas against intruder males and possibly to attract females. Vocalizations are low in frequency, predominately around 60-100 Hz, with some broadband components reaching up to 5 kHz, and range from 4-10 seconds in duration. Previously, source levels of vocalizations have only been estimated. Here we present a method for measuring vocalizations to estimate source levels of harbor seal roars. We used a three-element hydrophone array in Glacier Bay National Park to record harbor seal vocalizations during the breeding season (June—July 2015). Using the array, we localized individual vocalizing animals and calculated source levels of calls in the 40-500 Hz range. Knowledge of source levels for vocalizations of harbor seals is important for understanding how anthropogenic forces, such as noise from vessels, may impact the acoustic behavior and communication range of these animals.

4pABc14. Unsupervised clustering of toothed whale species from echolocation clicks. Yun Trinh, Scott Lindeneau (Comput. Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182, ytrinh@rohan.sdsu.edu), Maya Ackerman (Comput. Sci., San Jose State Univ., San Jose, CA), Simone Baumann-Pickering (Scripps Inst. of Oceanogr., La Jolla, CA), and Marie Roch (Comput. Sci., San Diego State Univ., San Diego, CA)

Supervised learning of categories related to animal signals requires labeled data that are used to train a classifier. In areas where species assemblages are poorly understood, such labeled data are usually unavailable. We show that unsupervised learning techniques may be used to provide an initial assessment of an area. We demonstrate that clustering techniques can be used to group sets of echolocation clicks from different species into clusters that tend to contain a single species without any labeled examples. Results are presented on the 2015 Detection Classification Localization and Density Estimation (DCLDE) toothed-whale development dataset, which contains analyst labels for several species of odontocetes as well as an unknown category. Echolocation clicks were detected and grouped into acoustic encounters from which spectral and temporal features were extracted. We used a simplifying assumption of single species encounters and estimated distributions of features in each encounter. Dendrograms were constructed by average-linkage clustering using a symmetric Kullback-Leibler similarity metric. Different algorithms were used to partition the dendrogram, with partition quality estimated by the silhouette algorithm. Comparison of the best machine-generated partitioning with that produced by analysts yielded an adjusted Rand statistic of 0.66, demonstrating a good degree of concurrence.

4pABc15. Monitoring of swine sneezing using time-frequency analysis for detecting diseases. Takuji Kawagishi (Graduate School of Systems and Information Eng., Univ. of Tsukuba, Tennodai1-1-1, Acoust. Lab. of Program in Intelligent Interaction Technologies, Tsukuba, Ibaraki 305-8571, Japan, kawagishi@aclab.esys.tsukuba.ac.jp), Koichi Mizutani, Keiichi Zempo, Naoto Wakatsuki (Faculty of Eng., Information and Systems, Univ. of Tsukuba, Tsukuba, Japan), Nobuhiro Takemae, and Takehiko Saito (Div. of Transboundary Animal Disease, National Inst. of Animal Health, NARO, Tsukuba, Japan)

This research mentions a detection method of swine sneezing in noisy swine house. While it is important to prevent swine infectious disease expansion in early stage, it is difficult to detect an infection in early stage because we cannot judge an infection from appearance of swine. However, it is a well-known fact that a swine with an infection tends to sneeze much more than healthy swine. The aim of this research is to achieve a detection of diseased swine in an automated way using the sound in the swine house and signal processing. We found that the sound of swine sneeze includes high frequency components 5-30 kHz and a sound feature of spectra, compared to the environmental noise in swine house. Based on these features of sneezing sound, we propose the detecting technique of sneezing sound based on the composition template of a frequency feature by a time-frequency analysis which is calculated from a lot of sneezing sounds. This method calculates the difference between frequency values of templates and recorded sounds. That could judge the recorded sounds as sneezing if sounds are smaller than the threshold. As a result, the method demonstrated effectiveness in a detection of swine sneezing.

4pABc16. Age-related hearing loss in TrpV1 knockout mice. Hongzhe Li (VA Loma Linda Healthcare System, 11201 Benton St., Res. Service, Loma Linda, CA 92357, Hongzhe.Li@va.gov)

Several transient receptor potential (TRP) channels including TRPV1 are located on hair cell membranes. The gating mechanisms of these channels are associated with cellular stress, inflammation, and cytoplasmic uptake of aminoglycosides. Thus, TRPV1 channel may serve as a functional target that links acoustic trauma and enhanced aminoglycoside trafficking. TrpV1 mutant mice have no apparent hearing loss as young adults. Here, we assessed the hearing sensitivity of TrpV1 mutant mice (B6.129X1-Trpv1^{tm1Jul}/J, stock #3770) by auditory brainstem responses to a broad frequency range, from 4 to 48 kHz. At 7 weeks of age, the auditory sensitivity in TrpV1 mutant mice was comparable to their heterozygous littermates. By 11 weeks of age, mutant TrpV1 mice exhibited evident high frequency hearing loss, with 30-to-40 dB elevated thresholds between 24 and 48 kHz, compared to littermate controls. At 18 weeks of age, high frequency hearing sensitivity worsened in both groups of mice, and this age-related progressive hearing loss was further expedited in TrpV1 mutant mice compared to littermate controls. Further research is required to determine the otoprotective role of TRPV1 channel in maintaining auditory sensitivity as mice age, and whether TrpV1 functionality exacerbates aminoglycoside-induced ototoxicity.

4pABc17. Demonstration of limited infrasonic sensitivity in the mallard duck (*Anas platyrhynchos*). Evan M. Hill (Psych., Univ. of Nebraska-Kearney, 2507 11th Ave., COPH - Psych., Kearney, NE 68849, hillem@unk.edu)

The purpose of this study was to determine the hearing range and sensitivity of the mallard duck, for the comparative purposes of identifying if the detection of infrasound (i.e., frequencies below 20 Hz) is a common feature of the avian auditory system. Prior to this work, only two species have had their low-frequency sensitivity fully assessed: the domestic pigeon (*Columba livia*), and the domestic chicken (*Gallus gallus domsticus*). Both species showed sensitivity to infrasonic acoustic signals as low as 2 Hz. The results of this study found the mallard to have a limited ability to detect infrasound, but inability to detect signals below 16 Hz. Given the lack of sensitivity to infrasound relative to what was observed in other species, this ability likely has limited functional value for this species. Future research on the topic should focus on developing a comprehensive theory that would allow for the selection of avian species for testing, to determine the origin and purpose of infrasonic sensitivity in birds.

4pABc18. Tissue physical property in the head of small toothed whales: Effect on the clicks propagation and directivity. Mika Kuroda (Hokkaido Univ., 3-1-1, Minato-cho, Hakodate, Hokkaido 041-8611, Japan, mika.kuroda@fish.hokudai.ac.jp), Motoki Sasaki (Obihiro Univ. of Agriculture and Veterinary Medicine, Obihiro, Japan), Kazutaka Yamada (Azabu Univ., Sagamihara, Japan), Nobuhiro Miki (Future Univ. Hakodate, Hakodate, Japan), Masao Amano (Nagasaki Univ., Nagasaki, Japan), Tadasu K. Yamada (National Museum of Nature and Sci., Tokyo, Tsukuba, Japan), and Takashi Matsuishi (Hokkaido Univ., Hakodate, Japan)

In the current study, the distribution of acoustic impedance in the head of harbor porpoise and striped dolphin were reported and the transmission factor in the head was calculated. Clicks properties are considered to be closely connected to the structure of sound producing organ, however, few effective investigations have been carried out to reveal the cause of the species diversity of clicks properties. Clicks of toothed whales has been confirmed to be produced at the dorsal barsae (DB) and propagated to the melon where the clicks' beam is progressively focused, and emitted into the seawater from the emitting surface (ES), a circular aperture at the frontal part of the melon. In our observation of the melon, a continuous gradients were observed from DB to ES for three species. At the ES, acoustic impedance of melon was matched with that of seawater. This mechanism would enable efficient click propagation and emission. The estimated directivity of clicks only from the transmission factor was much weaker than observed directivity, which suggests that the directivity of clicks is strongly affected by the refraction in the melon than a reflection and attenuation in the head tissue.

4pABc19. Cicada sound impacts avian dawn chorus in a subtropical forest of Taiwan. Yi C. Chen and Tzung S. Ding (School of Forestry and Resources Conservation, National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei, 10617 Taiwan, Taipei 10617, Taiwan, r04625003@ntu.edu.tw)

Acoustic space is a limited resource for animal communications. To reduce signal competition, animals could differentiate their acoustic frequency range or time to produce sound. Many birds call or sing at dawn for higher signal transmission efficiency. However, in tropical and subtropical forests, the dawn chorus were frequently interferenced by cicadas. In order to examine the effect of cicadas on the avian dawn chorus, we used automated acoustic recorders at subtropical forest of Yangmingshan National Park, Taiwan, to monitor acoustic signals of eight bird species and three cicada species. We found that all bird species postponed their acoustic

activities during cicada's breeding season. Meanwhile, we also found frequency partitioning of acoustic signals between birds and cicadas when their acoustic signals overlapped temporally. The results show that cicadas could play a crucial role in the avian dawn chorus. Birds would delay their signal producing time and acoustic frequency to partition the acoustic niche.

4pABc20. Variation in frequency-modulated call type for different flight densities in Mexican Free-tailed bats. Cassi Mardis, Stephanie Dreessen, and Laura Kloepper (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, cmardi01@saintmarys.edu)

Mexican free-tailed bats (*Tadarida brasiliensis*) adapt their call type depending on multiple factors. These bats have two general types of echolocation sounds, frequency-modulated (FM) and constant frequency (CF). CF calls are generally produced during the search phase of foraging, and FM calls are used during approach, prey capture, and roost emergence. In this study, we examined how Mexican free-tailed bats adapt their calls during emergence to avoid echo interference. We extracted individual calls from within a 500-ms sample that corresponded to five different emergence density categories. We found that these bats produce eight different FM call types during emergence. For each call within the sample, we characterized the call shape and calculated different parameters (duration, start frequency, peak frequency, end frequency). We compared the acoustic parameters across density categories and call shape, and compared the distribution of call shape across density categories. The results indicate that the call parameters and call shape vary from bat to bat, but there is no clear trend that this variance is due to the bat density. Therefore, these bats are likely changing their call shape and frequency characteristics to distinguish their own echoes from those of other bats.

4pABc21. The relationships between vocalizing bearded seals and sea ice extent in the southern Chukchi Sea. Mina Jimbo (Environ. Sci., Hokkaido Univ., Benten, 20-5, Hakodate, Hokkaido 040-0051, Japan, jimbo.min56@gmail.com), Daisuke Mizuguchi (Japan Fisheries Res. and Education Agency, Kushiro, Hokkaido, Japan), Koki Tsujii (Environ. Sci., Hokkaido Univ., Hakodate, Hokkaido, Japan), Kazushi Miyashita, and Yoko Mitani (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate, Hokkaido, Japan)

Bearded seal *Erignathus barbatus* is one of ice-obligate species, which breeds on seasonal sea-ice in the Arctic and sub-Arctic during March-May. They are threatened by the rapid changes in the Arctic environment due to recent climate change, which causes sea-ice reduction or transition of their prey biomass. To understand the seasonal distribution of bearded seals and assess the impacts of climate change, we deployed the underwater sound recorder at the Southern Chukchi Hotspot (SCH) (67.72^{o}N, 168.83^{o}W) during July 2012-October 2015. Calls of bearded seals were manually detected from September before sea-ice formation started, temporarily decreased during late-November-early-December when sea-ice were formed and increased again from January to the end of our recording periods (March 2013, May 2014 or June 2015) when the SCH were freezing. Combined with previous study, our results indicate that bearded seals come to the SCH from the northern part (i.e., Beaufort Sea) in September possibly to forage, and some move southerly to their breeding area (i.e., Bering Sea) coincident with sea-ice formation. These results suggest that seasonal distribution and vocal activity of bearded seals at the SCH, which are tied with sea-ice, might be changed by the Arctic sea-ice decline in the future.

4pABc22. Localization of acoustic windows in the beluga by acoustic delays from different sound-source positions. Evgeniya Sysueva, Alexander Supin, Vladimir Popov, Dmitry Nechaev (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, evgeniasysueva@gmail.com), Alena Lemazina (Lomonosov Moscow State Univ., Moscow, Russian Federation), and Mikhail Tarakanov (Inst. of Ecology and Evolution, Moscow, Russian Federation)

Acoustic windows were localized by acoustic delays from different sound-source positions in a beluga whale (Delphinapterus leucas). The delays were measured by recording the earlies component of the auditory evoked potential (AEPs) to short tone pips. This component recorded from the lateral head surface reflects the activity of the auditory nerve and features true monaural properties. Monaural nature of the component allows to localize an acoustic window with a better precision than binaural components. The acoustic delays were measured from a sound source at different positions at frequencies from 22.5 to 90 kHz (half-octave steps).A common sound-receiving region was identified for all these frequencies. The relation of the results to hypotheses of sound conduction in odontocetes is discussed. The use and care of the animals were adhered to the guidelines of the Russian Academy of Sciences for Research Involving Human and Non-Human Animals.

4pABc23. Intraspecific variation in the swimbladder occurrence and possibility of acoustic discrimination of the dominant mesopelagic fish *Diaphus garmanii*off western Kyushu, Japan. Hiroki Yasuma, Shinya Ohshima (Graduate School of Fisheries Sci., Hokkaido Univ., 3-1-1 Minato, Hakodate, Hokkaido 0418611, Japan, yasuma@fish.hokudai.ac.jp), Tohya Yasuda (Seikai National Fisheries Res. Inst., Nagasaki, Japan), Masa-aki Fukuwaka (Hokkaido National Fisheries Res. Inst., Hokkaido, Japan), and Koki Abe (National Res. Inst. of Fisheries Eng., Ibaraki, Japan)

Diaphus garmanii (Family, Mictophidae) is the most abundant mesopelagic fish in the continental slope off western Kyushu. For the application of acoustic monitoring, we have observed the swimbladder morphology of this species to estimate the target strength (TS). In this session, we show the intraspecific variation in the occurrences and shapes of swimbladder in *D. garmanii*, and propose the acoustic discrimination of the swimbladder morphology for accuracy improvement in biomass estimates. Acoustic data and biological samples were collected in the summer of 2012 to 2014. Both of fishes with (30% in all samples) and without (70%) swimbladder were observed by the soft X-ray method, and frequencies of the swimbladder occurrence were significantly different between male and female fishes. Theoretical scattering models showed that the TS difference between 38 kHz and 70 kHz was the most preferable parameter to discriminate the swimbladder occurrence of *D. garmanii*in field acoustic data. Field acoustic data analyses implied different patterns of vertical distribution between swimbladder and swimbladderless (male and female) fishes.

Session 4pAO

Acoustical Oceanography, Animal Bioacoustics, and Signal Processing in Acoustics: Acoustic Scattering by Aquatic Organisms II

Kelly J. Benoit-Bird, Cochair
College of Earth, Ocean, and Atmospheric Sciences, Oregon State University, 104 COEAS Admin Bldg., Corvallis, OR 97331

Kouichi Sawada, Cochair
Fisheries Technology, National Research Institute of Fisheries Engineering, FRA, 7620-7, Hasaki, Kamisu 3140408, Japan

Timothy K. Stanton, Cochair
Woods Hole Oceanographic Institution, Dept. Appl. Ocean. Phys. & Eng., Woods Hole Oceanographic Institution, Woods Hole, MA 02543

Contributed Papers

1:00

4pAO1. Three dimensional observation of pelagic fish schools by multi-beam echosounder ME70. Koki Abe, Tomohiko Matuura (National Res. Inst. of Fisheries Eng., Japan Fisheries Res. and Education Agency, 7620-7, Hasaki, Kamisu, Ibaraki 314-0408, Japan, abec@fra.affrc.go.jp), Tohya Yasuda, and Yohei Kawauchi (Seikai National Fisheries Res. Inst., Japan Fisheries Res. and Education Agency, Nagasaki, Nagasaki, Japan)

In this study, three-dimensional observation of pelagic fish using a multibeam echosounder with a quantitative echosounder are discussed. Multibeam echosounder originally has been developed for bathymetry survey, transmits and receives fan-shaped acoustic beam, and has extensive search area compared with single beam conventional echosounder. SIMRAD ME70 used in this study was developed specifically for fisheries research while maintaining the characteristics of fan-shaped wide multibeam. Because the calibration software is built in the ME70 for quantitative analysis of the acoustic beams, the output data can be compared with that from the quantitative echosounder like the EK60. Therefore, the ME70 and the EK60 are installed in the R/V YOKO-MARU used in alternate transmission, and carried out the observation for the same fish schools. Echograms were obtained from both echosounders and compared in terms of differences in characteristics of these echograms. Multibeam echosounder covered a widespread area to the side way of the vessel that the conventional echosounder could not observe and allowed to express fish schools in three dimensions.

1:15

4pAO2. Size estimation of a giant jellyfish *Nemopilema nomurai* using quantitative echosounder in the East China Sea. Tomohiko Matsuura, Koki Abe (Fisheries Technol., National Res. Inst. of Fisheries Eng., Hasaki 7620-7, firmheights #202, Kamisu-shi, Ibaraki 314-0408, Japan, mtsr@fra.affrc.go.jp), Haruya Yamada, and Yoko Kiyomoto (Fisheries management and Oceanogr., Seikai National Fisheries Res. Inst., Nagasaki, Japan)

The maximum bell-diameter of giant jellyfish *Nemopilema nomurai* is over one meter. They are transported by the Tsushima current to the Sea of Japan and are sufficient to seriously damage coastal fisheries. It is necessary to monitor the population and the distribution of them for reduction of the damage. Therefore, acoustic-optical surveys of the giant jellyfish have been conducted around the Tsushima Strait by a visual observation, a quantitative echosounder (EK60, SIMRAD) and an underwater video camera from 2009 to 2014. The giant jellyfish's size estimation method using an echotrace height was developed on the surveys. Using the method, the habitat and size distribution were obtained around the East China Sea in June and July 2015 to confirm the appearance of them earlier. The giant jellyfish were distributed from surface to about 40 m depth as results of the quantitative echosounder and the visual observation. The measured bell-diameter range was from 30 to 80 cm and giant jellyfish about 60 cm was most detected by the visual observation. The estimated bell-diameter range using the quantitative echosounder was from 10 to 80 cm. The value of highest frequency was 53 cm and agreed with the result of the visual observation.

1:30

4pAO3. Statistics of the backscattered returns from random aggregations of omnidirectional point scatterers and comparisons to fish schools measurements. Adaleena Mookerjee, David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., 2010 Autolab, Ann Arbor, MI 48109, adaleena@umich.edu), and Kelly Benoit-Bird (Monterey Bay Aquarium Res. Inst., Moss Landing, CA)

Remote assessment of scattering objects in the ocean by analyzing the statistics of backscattered returns has been of interest for decades. This presentation describes how simulated and measured backscatter results may be combined to assess the natural variability of fish density in extended schools. The simulations were based on numerical solution of Foldy's (1945) equations for harmonic illumination of spherical and spheroidal aggregations of randomly-placed omnidirectional point scatterers. The aggregations contained hundreds to thousands of scatterers, and varied in size from $12 \leq ka \leq 32$, where k is the acoustic wave number and a is the aggregation's volume-equivalent radius. Based on thousands of realizations, simulated backscattered returns were found to be Rayleigh distributed independently of the scatterers' and aggregations' characteristics. However, the distribution of volumetric scattering strengths from echo-sounder measurements from natural fish schools is broader. When taken together, the two results suggest that backscattering from natural aggregations of fish might be well described by a weighted superposition of Rayleigh distributions with the weighting function being a natural aggregation characteristic that can be remotely monitored. Results from simulations and measurements at 38 kHz and 120 kHz are shown and compared. [Work supported by ONR and UM Advanced Research Computing.]

1:45

4pAO4. Broadband target strength measurements of juvenile walleye pollock (*Theragra chalcogramma*). Kazuo Amakasu (Tokyo Univ. of Marine Sci. and Technol., 4-5-7 Konan, Minato-ku, Tokyo 108-8477, Japan, amakasu@kaiyodai.ac.jp), Tohru Mukai, Jun Yamamoto, Hiroki Hayashi, and Kohji Iida (Hokkaido Univ., Hokkaido, Japan)

Walleye pollock (*Theragra chalcogramma*) is one of very important fisheries resources in Japan. The acoustic-net surveys have been conducted around Hokkaido since 1995. To improve the present resource management, the estimation of the standing stock of juvenile walleye pollock is necessary. The target strengths of juvenile walleye pollock at 38 kHz have been well-studied through measurement. However, there are little knowledge of the target strengths at frequencies other than 38 kHz. In this study, therefore, the target strengths at other frequencies were measured. The acoustic system was consist of a broadband transducer and commercially available equipment. The calibration of the system was performed by using a 20.6-mm-diameter tungsten carbide sphere. The transmitting signal was 20-160 kHz linear frequency modulated signal. The target strengths of nine specimens were measured and their fork lengths were ranged from 6.9 to 12.1 cm. Each individual specimen was anesthetized and tethered in a tank (5 m x 5 m x 5 m) with a thin fishing line. Orientation angles of the specimen were varied between −50 to 50 degrees and the target strengths were measured at each angle. Orientation and length dependences of the target strengths are presented and discussed.

2:00

4pAO5. Modeling target strength of individual herring (Clupea harengus) at any aspect as a function of pressure and frequency. Geir Pedersen (Instrumentation, Christian Michelsen Res. AS, P.O. Box 6031, Bergen 5892, Norway, geir.pedersen@cmr.no), Gavin J. Macalay, Hector Peña (Inst. of Marine Res., Bergen, Norway), and Sascha M. Fässler (Wageningen Inst. for Marine Resources and Ecosystem Studies, Ijmuiden, Netherlands)

Fish target strength is primarily dependent on the physical dimensions of the fish, the acoustic frequency, and the orientation of the fish. In traditional vertically observing echosounder surveys, fish are insonified in the dorsal aspect with fairly limited tilt angle variation. In oblique-angled sonar surveys, however, fish may be insonified at other aspects. For herring, target strength is also depth dependent as they cannot refill the swimbladder at depth. Understanding the depth dependent target strength from several insonification angles is thus required for quantitative measurements with sonar. The dataset used in this study consists of seven herring, imaged using magnetic resonance imaging (MRI). The herring were placed in a pressure chamber inside the MRI, and subjected to different pressures corresponding to water depths of 0, 20, 40, and 60 m. Images were acquired of each specimen at each pressure. The swimbladders were segmented and 3D models of the swimbladders for each fish and pressure were constructed. These models were then used for computing the directivity pattern of the swimbladder at any angle as a function of frequency using the finite element method. Modeling results are also compared with measured dorsal and side aspect TS at different depths. The broadband backscattering (30-200 kHz) is evaluated for the potential for species and size discrimination and the effect including the fish body is discussed.

2:15–2:30 Break

2:30

4pAO6. Acoustic scattering by tube-building worms (Polychaeta: Maldanidae) of the New England Mud Patch. Matthew C. Zeh, Preston S. Wilson, Kevin M. Lee, Megan S. Ballard (Dept. of Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 204 East Dean Keeton St., Austin, TX 78712-1591, mzeh@utexas.edu), and Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin, AL)

The acoustic effects of biological activity within shallow water ocean environments are not well understood, yet are of increasing importance in some sonar applications. These environments remain insufficiently characterized in part due to the presence of benthic organisms within and near the sediment and water-sediment interface. Among the most prevalent infauna found within muddy ocean sediments are tube-building worms (Polychaeta: Maldanidae). The presence of and bioturbation caused by these organisms can affect the acoustical properties of the ocean bottom sediment. These effects are being studied in experiments being conducted in the New England Mud Patch, on the continental shelf south of Martha's Vinyard, MA. To begin to quantify these effects, laboratory acoustic scattering measurements of naturally collected worms and worm tubes were completed over a range of frequencies (50 kHz to 1 MHz) and incidence angles. Predictive models by Faran [J. Acoust. Soc. Am. **23**, 405 (1951)] approximating the worms as elastic cylinders, and Doolittle [J. Acoust. Soc. Am. **39**, 272 (1966)] approximating the worm tubes as cylindrical shells, were used to interpret the measured results. Measurements and model comparisons will be discussed along with implications relating to effective bulk sediment properties. [Work supported by ONR.]

2:45

4pAO7. Modelling macroalgae scattering (Ecklonia radiata). Jean-Pierre Hermand (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, av. F.D. Roosevelt 50, CP165/57, Brussels, Brussels Capital 1050, Belgium, jhermand@ulb.ac.be)

This paper will discuss the numerical modelling of acoustic scattering by macroalgae. Using earlier measurement of tissue acoustic properties, the scattering function of a thalus of *Ecklonia radiata* species is estimated by means of finite elements. Results will be presented for frequencies up to 20 kHz. [Work supported by ONR and ONR Global.]

3:00

4pAO8. An inter-individual communication biotelemetry system using Pseudo-random noise (Gold code). Yoshinori Minamoto (Tokyo Univ. of Marine Sci. and Technol., 4-5-7 Konan, Minato, Tokyo 108-8477, Japan, miyamoto@kaiyodai.ac.jp), Nobuaki Arai (Field Sci. Education and Res. Ctr., Kyoto Univ., Kyoto, Japan), Hiromichi Mitamura (Graduate School of Informatics, Kyoto Univ., Kyoto, Japan), Keiichi Unhid (Tokyo Univ. of Marine Sci. and Technol., Tokyo, Japan), Satoko S. Kimura (Field Sci. Education and Res. Ctr., Kyoto Univ., Kyoto, Japan), Kazushi Miyashita, Hokuto Shirakawa, Yuichi Tsuda (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate, Japan), Takashi Kitagawa (Atmosphere and Ocean Res. Inst., The Univ. of Tokyo, Kashiwa, Japan), and Minlee Yap (Tokyo Univ. of Marine Sci. and Technol., Tokyo, Japan)

Recently, bio-logging techniques involving electronic data-storage tags (data-logger) and acoustic transmitters have been increasingly used to understand migratory fish movements and behaviors. The monitoring of migratory fish can provide important insights into the marine ecosystem dynamics. The number of tags data-loggers used, however, is normally limited due to cost and the data-logger recovery rate is usually low. Therefore, we developed a new system that combines the acoustic communication and data logger. In other words, a data-logger that can share data with each other. One of the challenges is to develop an inter-individual communication biotelemetry system based on hydro-acoustic methods. In this system, we used the phase modulation of pseudo-random noise (Gold code). At the final stage of development, we will be doing conduct a field test of this inter-individual communication biotelemetry system on wild herrings and bonitos in the open sea. We expect this new technology to help overcome the bottleneck of the conventional bio-logging and bio-telemetry, and lead to a breakthrough in marine ecosystem studies.

3:15

4pAO9. Global access to sonar data: Where can it take you? Carrie C. Wall, Charles Anderson (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 216 UCB, Boulder, CO 80309, carrie.bell@colorado.edu), and Michael Jech (NEFSC, NMFS, Woods Hole, MA)

Scientific echosounders aboard NOAA fishery survey vessels are used to estimate biomass, measure fish school morphology, and characterize habitat. These surveys produce large volumes of data that are stored locally and difficult to access. Data that are easily discoverable and accessible provide valuable information beyond their original collection purpose. NOAA's National Centers for Environmental Information, in partnership with the National Marine Fisheries Service and the University of Colorado, created a national archive for the stewardship and distribution of water column sonar data. A data access web page allows users to query the metadata and access the raw sonar data. Visualization products allow researchers and the public to understand the quality and content of large volumes of archived data. Such products transform the complex raw data into a digestible image and are highly valuable for a broad audience of varying backgrounds. Concurrently collected oceanographic and bathymetric data are being integrated into the data access page to provide an ecosystem-wide understanding of the area surveyed. Benefits of the archive include global access to an unprecedented nationwide dataset and the increased potential for researchers to address new questions to advance the field of marine ecosystem acoustics.

THURSDAY AFTERNOON, 1 DECEMBER 2016 CORAL 1, 1:00 P.M. TO 5:30 P.M.

Session 4pBA

Biomedical Acoustics: Medical Acoustics in Kidney and Liver Disease II

Norihiro Koizumi, Cochair
Graduate School of Informatics and Engineering, The University of Electro-Communications (UEC), 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585 JAPAN, Chofu 182-8585, Japan

Michael Bailey, Cochair
Center for Industrial and Medical Ultrasound, Applied Physics Lab, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Hiroyuki Fukuda, Cochair
Yokohama City Univ. Medical Center, Yokohama, Japan

Adam D. Maxwell, Cochair
University of Washington, 1013 NE 40th St., Seattle, WA 98105

Contributed Papers

1:00

4pBA1. Kidney stone pushing and trapping using focused ultrasound beams of different structure. Oleg A. Sapozhnikov, Anastasia V. Nikolaeva, Marina E. Terzi, Sergey A. Tsysar (Phys. Faculty, Moscow State Univ., Leninskie Gory, Phys. Faculty, Moscow State University, Moscow 119991, Russian Federation, oleg@acs366.phys.msu.ru), Adam D. Maxwell, Bryan W. Cunitz, and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

A technology to reposition kidney stones with radiation force was recently proposed by our team and already used to transcutaneously facilitate passage of small stones. While successful, the trial revealed a need for optimization of the ultrasound beam structure, frequency, and intensity to make it more effective. In the current work, the effect of the ultrasonic beam diameter vs. stone size using a quasi-Gaussian beam model was numerically investigated. Radiation force on a kidney stone was found to be strongest when the beam width was slightly wider than the stone diameter. This can be explained by more effective generation of shear waves inside the stone resulting from their effective coupling with the acoustic waves in liquid at the stone edges. In another study, the possibility of using vortex beams to trap the stones in the lateral direction was investigated. Both theoretical modeling and experiments were performed using two systems: a single-element transducer combined with a sector-shaped phase plate and a 12-element sector array. Human stones approximately 3-5 mm, as well as glass and styrofoam beads, were controllably translated along the surface transverse to the beam. [This work was supported by RBBR 14-02-00426, NIH NIDDK DK43881, DK104854, and DK092197, and NSBRI through NASA NCC 9-58.]

1:15

4pBA2. Robust servoing method for renal stones/tumors for the non-invasive ultrasound theragnostic system. Atsushi Kayasuga (The Univ. of Tokyo, 8-14-6, Goko, Matsudo-shi, Chiba 270-2213, Japan, kayasuga04@gmail.com), Norihiro Koizumi, Kyohei Tomita, Yu Nishiyama (The Univ. of Electro-Communications, Tokyo, Japan), Hiroyuki Tsukihara (The Univ. of Tokyo, Tokyo, Japan), Hiroyuki Fukuda (The Univ. of Yokohama City, Tokyo, Japan), Kiyoshi Yoshinaka (National Inst. of Adv. Industrial Sci. and Technol., Tokyo, Japan), Takashi Azuma, Hideyo Miyazaki, Naohiko Sugita (The Univ. of Tokyo, Tokyo, Japan), Kazushi Numata (The Univ. of Yokohama City, Tokyo, Japan), Yukio Honma, Yoichiro Matsumoto, and Mamoru Mitsuishi (The Univ. of Tokyo, Tokyo, Japan)

The main problem on HIFU (High Intensity Focused Ultrasound) therapy is the difficulty to locate HIFU focus precisely onto the focal lesion, which is located in the moving organ such as livers/kidneys, due to the deformation and rotation, which is caused by respiration. Furthermore, rib bones frequently block the acoustic path to the lesion. The acoustic shadow, which is generated by the rib bone, is observed in the ultrasound images and the lesion is hidden in the shadow. To cope with this problem, we have developed a novel method to track, follow, and monitor the lesion by utilizing the contour information of the organ, which incorporate the lesion under those difficult conditions. As for the tracking method, the contour of the organ, which is deformed and rotated in accordance with respiration, is extracted automatically in 2-D ultrasound images. The missing contour information in the acoustic shadow area is estimated and compensated by the surrounding contours, which are successfully extracted. To confirm the effectiveness of the proposed method, we compared the proposed method

with the conventional method in terms of the tracking performance of the stone. As a result, we achieved the tracking performance (the average servo error is 2.3 mm) by the proposed method, while the average servo error is 3.57 mm by the conventional method. This shows the validity of our novel proposed method.

1:30

4pBA3. Combining burst wave lithotripsy and ultrasonic propulsion for enhanced kidney stone comminution. Theresa A. Zwaschka, Bryan W. Cunitz, Michael R. Bailey, Barbrina Dunmire (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, azwaschka@ucla.edu), and Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA)

Burst wave lithotripsy (BWL) and ultrasonic propulsion (UP) are two new focused ultrasound technologies for noninvasively treating kidney stones. BWL applies short, focused bursts of ultrasound to fracture stones. UP employs long-duration bursts of ultrasound to reposition stones and has been successfully tested on humans. We hypothesize that application of low-level UP can improve comminution by reorienting the stone and dislodging intervening fragments from the stone surface. Experiments were performed in a degassed water bath with a polyvinyl chloride tissue phantom mimicking a kidney calyx. An artificial calcite stone was placed in the phantom and aligned with the focus of a 330-kHz BWL transducer with a co-axial P4-2 imaging probe used for UP. Fragmentation size was compared between stones treated with a fixed 5-minute BWL exposure using 20-cycle bursts at 40 bursts/sec, with or without interleaved UP. The results showed improved fragmentation with the combined exposure, with 3 UP bursts/minute resulting in 45% of stone debris < 2mm versus 23% without UP ($p = 0.011$). Stones treated with UP alone did not fragment. These results suggest UP can improve stone fragmentation by BWL. [Work supported by NIH K01 DK104854 and P01 DK043881, and NSBRI through NASA NCC 9-58.]

1:45

4pBA4. Clinical experience of intra-operative high intensity focused ultrasound in patients with colorectal liver metastases. Results of a Phase II study. David Melodelima (LabTAU - INSERM U1032, 151 cours Albert Thomas, Lyon 69003, France, David.Melodelima@inserm.fr), Aurelien Dupre, David Perol, Yao Chen (Dept. of Surgery, Ctr. Leon Berard, Lyon, France), Jeremy Vincenot, Anthony Kocot (LabTAU - INSERM U1032, Lyon, France), and Michel Rivoire (Dept. of Surgery, Ctr. Leon Berard, Lyon, France)

The aim of this study was to assess the feasibility, safety and accuracy of HIFU ablation in patients with liver metastases in a prospective, phase I-II trial. The transducer has a toroidal shape (diameter: 70 mm, radius of curvature: 70 mm) and was divided into 32 ring-shaped emitters operating at 3 MHz. Twenty-eight patients were included. HIFU ablations were created to ablate metastases (20 mm maximal diameter) with safety margins in all directions. The exposure time varied from 40 seconds to 370 seconds according to the diameter of the metastases to be treated. One metastasis of 10 mm in diameter was ablated in 40 seconds with safety margins. Using electronic focusing metastases of 2 cm in diameter were ablated with safety margins ($>$ 3 mm in all directions) in 370 seconds. The dimensions of these HIFU ablations were a diameter of 48 $\pm$ 4.9 mm and a long axis of 51 $\pm$ 3.4 mm. No damage occurred to neighboring tissues. This HIFU device safely achieved ablations of small metastases ($<$ 20 mm) with planned safety margins of at least 3 mm in all directions. This study is the first clinical use of intra-operative HIFU in patients with liver metastases.

2:00

4pBA5. Generation of different types of surface acoustic waves by shock wave-Stone interaction. Ying Zhang, Chen Yang, and Pei Zhong (Mech. Eng. and Material Sci., Duke Univ., Hudson 229, Durham, NC 27708, zhang.ying@duke.edu)

The generation of different types of surface acoustic waves (SAWs) in lithotripsy is investigated by numerical simulations using COMSOL, considering either the case of a weakly focused shock wave as an incident plane wave (in the focal plane) in shock wave lithotripsy (SWL) or a spherically divergent shock wave produced by a spark discharge in Nano Pulse Lithotripsy (NPL). The interaction of these two types of shock waves with an artificial kidney stone immersed in water may generate three different types of SAWs: namely, Scholte wave, acoustic surface evanescent wave (ASEW), and leaky Rayleigh wave (LRW). In particular, we examined the generation of SAWs on a flat, cylindrical, or spherical surface under different incident angles. It was found that the geometry and acoustic properties of the stone could significantly influence the types of SAWs generated and the resultant peak tensile stress produced at the stone-water boundary. Comparison with experimental measurements (i.e., Schlieren and photoelastic imaging) and stone fracture will also be made. [Work supported by NIH through 5R37DK052985-20.]

2:15

4pBA6. Modeling and experimental analysis of acoustic cavitation bubble clouds for burst-wave lithotripsy. Kazuki Maeda, Tim Colonius (California Inst. of Technol., 1200 E California Blvd., Pasadena, CA 91125, maeda@caltech.edu), Wayne Kreider, Adam Maxwell (Univ. of Washington, Seattle, WA), and Michael Bailey (Univ. of Washington, Seattle, CA)

Understanding the dynamics of cavitation bubble clouds formed inside a human body is critical for the design of burst-wave lithotripsy (BWL), a newly proposed method that uses focused ultrasound pulses with amplitude of $O(10)$ MPa and frequency of $O(0.1)$ MHz to fragment kidney stones. We present modeling and three-dimensional direct numerical simulations of interactions between bubble clouds and ultrasound pulses in water. We study two configurations: isolated clouds in a free field, and clouds near a rigid surface. In the modeling, we solve for the bubble radius evolution and continuous flow field using a WENO-based compressible flow solver. In the solver, Lagrangian bubbles are coupled with the continuous phase, defined on an Eulerian grid, at the sub-grid scale using volume averaging techniques. Correlations between the initial void fraction and the maximum collapse pressure in the cloud are discussed. We demonstrate acoustic imaging of the bubbles by post-processing simulated pressure signals at particular sensor locations indicating waves scattered by the clouds. Finally, we compare the simulation results with experimental results including high-speed imaging and hydrophone measurements. The time evolution of the cloud void fraction and the scattered acoustic field in the simulation agree with the experimental results. [Funding supported by NIH 2P01-DK043881.]

Invited Paper

2:30

4pBA7. A training phantom for ultrasound-guided boiling histotripsy therapy. Barbrina Dunmire, John C. Kucewicz, Wayne Kreider (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, mrbean@uw.edu), Tanya D. Khokhlova (Medicine - Gastroenterology, Univ. of Washington, Seattle, WA), George R. Schade, Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Appl. Phys. Lab, Univ. of Washington, Seattle, Washington, U.S. and Dept. of Acoust., Moscow State Univ., Seattle, Washington), Larry A. Crum (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Vera A. Khokhlova (Appl. Phys. Lab, Univ. of Washington, Seattle, Washington, U.S. and Dept. of Acoust., Moscow State Univ., Seattle, Washington)

Boiling histotripsy (BH) uses millisecond-long focused ultrasound pulses with shocks to mechanically disrupt targeted tissue under real-time ultrasound monitoring. However, adipose tissue and ribs can interfere with BH therapy through aberration, absorption, and diffraction. Here we introduce a robust abdominal wall phantom that includes fat, muscle, and rib layers for demonstrating the use of BH and investigating the impact of anatomic structures on treatment success. The skin is a silicone sheet; the fat and muscle layers are polyvinyl alcohol phantoms with irregular-shaped walls; the ribs are 3D-printed sections from a human model anatomically relevant to liver or kidney treatments. The target is a transparent alginate or polyacrylamide gel that allows visualization of the lesion. The pieces are assembled in a water-filled container providing coupling between layers and allowing components to be shifted in position relative to the transducer. A BH transducer (1.2 MHz, 12.5 cm focal length, f# = 1) containing an imaging probe in its central opening was used for initial phantom assessment. Preliminary results show that each layer impacts the beam in ways similar to observations from *in vivo* and *ex vivo* experiments. [Work supported by NSBRI through NASA NCC 9-58, NIH RO1EB007643, K01EB015745, and K01DK104854.]

Contributed Paper

2:45

4pBA8. Broad beam for more effective ultrasonic propulsion of kidney stones. Michael R. Bailey, Bryan W. Cunitz, Barbrina Dunmire, Brian MacConaghy, Yak-Nam Wang (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, mike.bailey.apl@gmail.com), Karmon Janssen, Timothy C. Brand (Dept. of Urology, Madigan Health Care System, Tacoma, WA), Doug Corl, Oren Levy (SonoMotion, Inc., San Francisco, CA), Mathew D. Sorensen (Dept. of Urology, VA Puget Sound Health Care System, Seattle, WA), Jonathan D. Harper, and Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA)

Ultrasonic propulsion uses focused bursts of ultrasound to generate radiation force to non-invasively reposition stones. Ultrasonic propulsion has been implemented on a C5-2 curvilinear array and shown to be safe and effective for expelling small stones and fragments in a human feasibility study. Here, we evaluated the efficacy of three different focal beam patterns in moving stones: 1) the existing approach exciting all 128 elements of the C5-2 transducer at 2 MHz, 2) exciting only 40 elements, and 3) implementing a separate single element transducer at 350 kHz. The capability of each method to lift clusters of 1-2 mm and 3-4 mm calcium oxalate stone fragments in a pipette was measured, filmed and compared by calculating the sum of the product of fragment area and distance moved for all fragments. For the same peak pressure, the alternative approaches (2 and 3) lifted greater stone mass than the original output. With 1-2 mm fragments, improvement with the three methods was 14% and 500%. For 3-4 mm, improvement was 45% and 178%. [Work supported by NIH NIDDK grants DK043881, DK107094, DK104854, and DK092197, the NSBRI through NASA NCC 9-58, and resources from the VA Puget Sound Health Care System.]

3:00–3:15 Break

Invited Paper

3:15

4pBA9. HIFU-induced cloud cavitation control for kidney stone treatment. Teiichiro Ikeda (Res. and Development Group, Hitachi Ltd., 1-280, Higashi-koigakubo, Kokubunji, Tokyo 1858601, Japan, teiichiro.ikeda.hv@hitachi.com), Shin Yoshizawa (Dept. of Elec. and Commun. Eng., Tohoku Univ., Miyagi, Japan), and Yoichiro Matsumoto (RIKEN, Saitama, Japan)

The shock wave lithotripter uses an order of microsecond pulse durations and up to a 100 MPa pressure spike triggered at approximately 0.5—2 Hz to fragment kidney stones through mechanical mechanisms. One important mechanism is cavitation. We proposed an alternative type of lithotripsy method that maximizes cavitation activity to disintegrate kidney stones using high-intensity focused ultrasound (HIFU). We designed a two-frequency wave (cavitation control (C-C) waveform); a high-frequency ultrasound pulse to create a cavitation cloud, and a low-frequency pulse following the high-frequency pulse to force the cloud into collapse. High-speed photography showed cavitation collapse on a kidney stone and shock wave emission from the cloud. We also conducted *in-vitro* erosion tests of model and natural stones. For the model stones, the erosion rate of the C-C waveform showed a distinct advantage with the combined high- and low-frequency waves over either wave alone. Natural stones were eroded and most of the resulting fragments were less than 1 mm in diameter. The small fragments were small enough to pass through the urethra. The results demonstrate that, with the precise control of cavitation activity, focused ultrasound has the potential to be used to develop a less invasive and more controllable lithotripsy system.

Contributed Paper

3:30

4pBA10. Liver tracking system utilizing template matching and energy function in high intensity focused ultrasound/radio frequency ablation therapy. Kyohei Tomita, Norihiro Koizumi, Ryosuke Kondo (Graduate School of Informatics and Eng., The Univ. of Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan, tomitakyohei@uec.ac.jp), Atsushi Kayasuga (The Univ. of Tokyo, Bunkyo-ku, Japan), Yu Nishiyama (Graduate school of Informatics and Engineerling, The Univ. of Electro-Communications, Chofu, Japan), Hiroyuki Tsukihara (The Univ. of Tokyo, Bunkyo-ku, Jersey), Hiroyuki Fukuda, Kazushi Numata (Yokohama City Univ., Yokohama, Japan), Yoichiro Matsumoto (RIKEN, Wako, Japan), and Mamoru Mitsuishi (The Univ. of Tokyo, Bunkyo-ku, Japan)

Monitoring and evaluating the therapeutic effects, in ultrasound images during HIFU (High Intensity Focused Ultrasound) and RFA (Radio Frequency Ablation) therapies, are important. However, the common problem is the difficulty to monitor the progress and identify the positions of the focal lesion precisely in accordance with the progress of the treatment. This problem is caused by the movement of organs and the change of the textures information in ultrasound images. To overcome those problems, we have developed a novel method to track, follow, and monitor the focal lesion by the combination of the energy function method and the template matching method. The template matching method uses the characteristic texture information of the organ near the focal lesion. The energy function is implemented in order to reinforce the robustness of the tracking performance of the template matching method. Particularly, we evaluate the existing probability of the focal lesion considering the continuity of the movement and the relative distance from the surface of the liver. The estimated position of the focal lesion is obtained by minimizing the above mentioned energy function. The effectiveness of the proposed method is confirmed by comparing the novel proposed method and the conventional template matching method in terms of the tracking performance for the liver tumor during the real RFA treatments.

Invited Paper

3:45

4pBA11. Construction methodology for non-invasive ultrasound theragnostic system by medical digitalization. Norihiro Koizumi (Graduate School of Informatics and Eng., The Univ. of Electro-Communications (UEC), 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan, nkoizumi@ieee.org), Hiroyuki Tsukihara (The Univ. of Tokyo, Bunkyo-ku, Japan), Kiyoshi Yoshinaka (National Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Japan), Hideyo Miyazaki (The Univ. of Tokyo, Bunkyo-ku, Japan), Hiroyuki Fukuda (Yokohama City Univ., Yokohama, Japan), Kazushi Numata (The Univ. of Tokyo, Yokohama, Japan), Yukio Homma (The Univ. of Tokyo, Bunkyo-ku, Japan), Yoichiro Matsumoto (RIKEN, Bunkyo-ku, Japan), and Mamoru Mitsuishi (The Univ. of Tokyo, Bunkyo-ku, Japan)

We propose a noninvasive ultrasound theragnostic system (NIUTS) that compensates for movement by tracking and following the area to be treated by using stereo ultrasound imaging while irradiating the focal lesion with HIFU. Theragnostics involves therapeutics and diagnostics. Our proposed system uses focused ultrasound to destroy tumors and stones without damaging the surrounding healthy tissue. This is achieved by tracking and following the focal lesion (stones/tumors), in order to compensate for movement due to respiration, heartbeat, etc. In the overall research project, we aim to enhance the focal lesion servo (FLS) performances based on our original medical support system construction methodology, which is called "Me-DigIT (medical digitalization by IT technology, which includes especially robot technology)." In the present report, we illustrate the mechanisms, controllers, and robot vision core technology to enhance the focal lesion tracking and following performance of the NIUTS. As a result, with the constructed NIUTS, tracking performance within 2.5 mm was achieved for a healthy human kidney. The standard deviation of the position of the servo target was 1.92 mm with FLS, while it was 13.2 mm without FLS. In other words, 86 % of body movement could be compensated for by using NIUTS.

Contributed Papers

4:00

4pBA12. An ultrasound guided monitoring system for high intensity focused ultrasound and radio frequency ablation therapies. Ryosuke Kondo, Norihiro Koizumi, Kyohei Tomita (Graduate School of Informatics and Eng., The Univ. of Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan, k1312058@edu.cc.uec.ac.jp), Atsushi Kayasuga (The Univ. of Tokyo, Bunkyo-ku, Japan), Yu Nishiyama (Graduate school of Informatics and Engineerling, The Univ. of Electro-Communications, Chofu, Japan), Hiroyuki Tsukihara (The Univ. of Tokyo, Bunkyo-Ku, Japan), Hiroyuki Fukuda, Kazushi Numata (Yokohama city Univ., Yokohama, Japan), Yoichiro Matsumoto (RIKEN, Wako, Japan), and Mamoru Mitsuishi (The Univ. of Tokyo, Bunkyo-ku, Japan)

In accordance with the progress of the HIFU (High Intensity Focused Ultrasound) and RFA (Radio Frequency Ablation) treatments, the intensity of the focal lesion and the surrounding marginal area changes little by little. Here, it should be noted that the human eyes are very weak when the image takes a long time to change even if the image change from the start of the ablation treatment is large. To cope with those problems, we propose an ultrasound guided monitoring system to evaluate the progress of the ablation therapy quantitatively in order to secure the certain level of the monitoring during the HIFU and RFA therapies by reinforcing the lack of experience of medical doctors. Particularly, we propose a system to monitor and evaluate the intensity of the focal lesion and the intensity of the surrounding marginal area by tracking the position of the characteristic texture area near the focal lesion, which moves in accordance with the respiration. Our method to overlay the focal lesion and the surrounding marginal area on the ultrasound image is also effective in order not to miss the area to be treated. The experimental results during RFA ablation therapy shows the effectiveness of our constructed system.

4:15

4pBA13. Modeling shock-wave fields generated by a diagnostic-type transducer. Maria M. Karzova, Pavel B. Rosnitskiy, Petr V. Yuldashev, Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow 119991, Russian Federation, masha@acs366.phys.msu.ru), Wayne Kreider, Bryan W. Cunitz, Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Vera A. Khokhlova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

New applications of ultrasound imaging may benefit from increased *in situ* pressure levels. However strong nonlinear propagation effects have not been studied in detail for fields generated by diagnostic transducers. Here we compare two modeling approaches for predicting shock formation in fields generated by a curvilinear imaging probe (C5-2). The field was simulated in two ways: 1) a 3D full-diffraction model based on the Westervelt equation, with a boundary condition defined on a cylindrical surface; 2) an axially symmetric parabolic model based on the KZK equation to define a flat, circular equivalent source. For both models, boundary conditions are adjusted to match low-power axial pressure measurements in the focal lobe of the beam. Simulations and measurements were performed for operation of 40, 64, and 128 central probe elements, and a wide range of clinically relevant output levels was considered. Comparison of focal waveforms shows that the KZK model can predict the amplitudes of fully developed shocks within 5% for 64 and 128 active elements, and within 15% for 40 elements. The full 3D calculations provide better agreement with experiments (within 3%) but require significantly more computational resources. [Work supported by RSF 14-12-00974 and a scholarship of the president of Russia.]

4:30

4pBA14. Scattered shear wave contribution to acoustic radiation force on spheres in soft tissue. Benjamin C. Treweek, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, btreweek@utexas.edu)

A theory for acoustic radiation force on a sphere in soft tissue was developed for arbitrary incident compressional wave fields [Ilinskii *et al.*, Proc. Meet. Acoust. **19**, 045004 (2013)]. This theory includes two contributions to the radiation force. The first depends only on the incident and scattered compressional waves, whereas the second depends on the scattered shear waves as well. Each contribution in turn has two parts, one due to direct integration of the time-averaged Piola-Kirchhoff stress tensor over the surface of the sphere, and the other due to the irrotational component of the body force on the sphere. While both parts are known analytically for the compressional waves, only the first part has been obtained analytically for the contribution involving shear waves. The irrotational portion associated with shear waves and its effect on the total radiation force is the subject of this presentation. The analysis is conducted via Helmholtz decomposition of the body force associated with shear waves and subsequent integration of the irrotational portion over the surface of the sphere. Simplifying analytical approximations based on numerical calculations are examined for various elastic properties of the sphere and soft tissue. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

4:45

4pBA15. Thresholds for sustained bubble cloud generation in burst wave lithotripsy. Christopher Hunter (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Bryan W. Cunitz, Barbrina Dunmire, and Wayne Kreider (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA, wkreider@uw.edu)

Burst wave lithotripsy (BWL) is a new non-invasive approach for disintegrating kidney stones using tone bursts of sub-megahertz ultrasound. An important advantage of BWL compared to shock wave lithotripsy (SWL) is the rate at which acoustic energy can be delivered without causing cavitation that injures functional renal tissue. BWL treatments capable of breaking stones have been safely delivered to pig kidneys *in vivo* at a 40 Hz burst repetition frequency, representing a rate of energy delivery well in excess of that used in SWL. To facilitate the design of BWL treatments that safely and optimally break stones, this work focuses on understanding how treatment parameters affect the generation of sustained bubble clouds. High-speed photography and ultrasound imaging were used to characterize the presence and persistence of cavitation activity *in vitro*. Experiments were conducted for BWL treatments at 335 kHz to evaluate the impact of treatment rate, dissolved gas concentration, volumetric confinement (representing kidney collecting space), and presence/absence of a stone. Threshold pressures for generating cavitation clouds were observed to vary with each of these variables. [Funding support by NIH P01-DK043881, R01-DK092197, K01-DK104854, and NSBRI through NASA NCC 9-58.]

5:00

4pBA16. Evaluation of a model-based poroelastography algorithm for edema quantification. John J. Pitre (Biomedical Eng., Univ. of Michigan, 2125 Lurie Biomedical Eng. Bldg., 1101 Beal Ave., Ann Arbor, MI 48109, jpitre@umich.edu), William F. Weitzel (Dept. of Veterans Affairs Medical Ctr., Ann Arbor, MI), and Joseph L. Bull (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

A critical component of end stage renal disease treatment is optimal fluid management. Patients with reduced kidney function risk developing fluid overload, and many dialysis patients exceed recommended levels of fluid retention, leading to frequent clinical intervention and higher mortality. Despite this, current clinical standards of care rely on reactive semi-quantitative tests to grade fluid overload and peripheral edema. Poroelastography has been proposed as a quantitative means of estimating fluid overload in peripheral edema, but clinical studies have met limited success, in part because of measurement noise. We developed a new poroelastography method based on an inverse problem formulation to address the shortcomings of current methods. Our technique iteratively minimizes the error between ultrasound displacement measurements and the solution of a Biot poroelastic model. This formulation does not depend on noise-amplifying derivatives and ratios. We tested our algorithm in a simulation study using synthetic displacement data with Gaussian noise and speckle tracking measurements from simulated sonograms. Our algorithm accurately estimated the Poisson's ratio at all noise levels tested. Reconstruction errors for all cases were less than 10% and outperformed traditional methods. Although our method required 4-12 hours to run, it is easily parallelized. Future work will focus on benchtop validation.

5:15

4pBA17. Computer-aided B-mode ultrasound diagnosis of hepatic steatosis. Gert Weijers, Johan Thijssen (Radiology and Nuclear Medicine, Radboud Univ. Medical Ctr., Nijmegen, Netherlands), Geert Wanten, Joost Drenth (Gastroenterology, Radboud Univ. Medical Ctr., Nijmegen, N/A, Netherlands), Marinette van der Graaf, and Chris L. de Korte (Radiology and Nuclear Medicine, Radboud Univ. Medical Ctr., MUSIC 766, PO Box 9101, Nijmegen 6500HB, Netherlands, chris.dekorte@radboudumc.nl)

A computer aided ultrasound (CAUS) protocol was developed for the diagnosis and staging of hepatic steatosis. Calibration of the fixed imaging preset enables expression of the US parameters relative to the tissue mimicking phantom used and consequently a vendor independent comparison. The CAUS protocol was validated on high yielding dairy cows ($n = 151$), which severed as a reference model for human non-alcoholic fatty liver disease. The study showed high values for: area under the curve (A_{ROC}): 0.94; sensitivity: 87%; specificity: 83%. A pilot study in patients on Home Parenteral Nutrition (HPN, $n = 14$) and a more extensive study in an obese cohort ($n = 116$) was performed. Validation was performed by Magnetic Resonance Spectroscopy (MRS). Both studies showed that of all parameters, the residual attenuation coefficient (RAC, the depth dependent mean echo level corrected for beam profile and normal attenuation in liver tissue) has the highest correlation to the reference (MRS fat percentage). For the HPN cohort, a correlation of 0.91 was found. In the obese population, this value was only 0.64; however, an A_{ROC} of 0.86, a sensitivity of 75%, and a specificity of 84% were obtained. In conclusion, CAUS shows promising results for non-invasive quantification of liver steatosis.

Session 4pEA

Engineering Acoustics: Sound Field Control Techniques

Stephen Elliott, Cochair
Inst. of Sound and Vibration Res., Univ. of Southampton, Southampton, United Kingdom

Yoshinobu Kajikawa, Cochair
Kansai University, 3-3-35, Yamate-cho, Suita, Osaka 564-8680, Japan

Invited Papers

1:00

4pEA1. Controlling 3D sound field generated by a 2D surface from its 1D boundary. Xiaojun Qiu (School of Elec., Mech., and Mechatronic System, Univ. of Technol. Sydney, Sydney, NSW 2007, Australia, xiaojun.qiu@uts.edu.au), Shuping Wang, and Jiancheng Tao (Key Lab. of Modern Acoust., Inst. of Acoust., Nanjing Univ., Nanjing, China)

As described by the Kirchhoff-Helmholtz integral equation, for a 3D space without internal sources, the inside sound field is completely determined by the sound pressure and its normal gradient on the boundary, so if only a portion of the boundary surface vibrates as the primary sound source while all the remaining portion of the surface is acoustically rigid, a sufficiently large number of secondary sound sources can be located evenly on the primary sound surface to control the 3D sound field completely. The question to be investigated in this research is whether it is possible to control the 3D sound field by only applying secondary sound sources along the 1D boundary of the 2D primary sound surface, for example, controlling the sound transmission from a door with secondary sound sources only on its frame. Simulation and experimental results will be presented to demonstrate the feasibility of the idea.

1:20

4pEA2. Analysis and synthesis of spatial sound fields with room response modeling using an integrated microphone and loudspeaker array system. Mingsian R. Bai, Yi Li, and Yi-Hao Chiang (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan 30013, Taiwan, msbai63@gmail.com)

A unified framework is proposed for analysis and synthesis of spatial sound fields. In the sound field analysis (SFA) phase, an unbaffled 24-element circular microphone array (CMA) is utilized to “encode” the sound field based on the plane-wave decomposition, whereas in the sound field synthesis (SFS) phase a 32-element rectangular loudspeaker array is employed to “decode” the target sound field using pressure matching technique. Depending on the sparsity of the sound sources, the SFA stage can be implemented in two ways. For the sparse-source scenario, a two-stage algorithm is utilized to estimate the source bearings using the minimum power distortionless response (MPDR) and the associated amplitude coefficients of plane waves using the Tikhonov regularization (TIKR) algorithm. Alternatively, a one-stage algorithm based on compressive sensing (CS) algorithm can be used. For the nonsparse-source scenario, a one-stage TIKR is utilized to solve for the amplitude coefficients for plane-wave components uniformly distributed in the angular domain. The SFA technique for the non-sparse source scenario is also useful in establishing the room response model, as required in the pressure matching step of the SFS phase. The integrated acoustic array system is validated with localization and listening tests.

1:40

4pEA3. Three-dimensional sound field reproduction based on spatial fusion of carrier and sideband waves with parametric array loudspeaker. Masato Nakayama and Takanobu Nishiura (College of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Noji Higashi, Kusatsu, Shiga 525-8577, Japan, mnaka@fc.ritsumei.ac.jp)

It is very important to provide a personal audible space (audio-spot) to listeners. Parametric array loudspeaker (PAL) has been proposed in order to provide an audio-spot because it has super directivity. It however has difficulty in reproducing the audible sound only to the particular space because the acoustic beam with the PAL is reflected and intercepted. Therefore, we have proposed the three-dimensional sound field reproduction based on spatial fusion of carrier and sideband waves with the PAL. In addition, we have proposed the method of near-sound-field reproduction based on spatial fusion of the carrier and sideband waves. In this presentation, we explain principle of the proposed method, and its applications. In the proposed method, we utilize principle which the intense ultrasound self-demodulates to the audible sound by nonlinear interaction in the air. More specifically, the intense ultrasound consists of the carrier and sideband waves, and the audible sound is demodulated as the different tone between carrier and sideband waves. The proposed method forms the audio-spot by fusing the carrier and sideband waves in the particular space. Finally, we evaluate the effectiveness of the proposed method through the evaluation experiments.

2:00

4pEA4. Perceptual optimization of sound field reproduction system and sound field rendering with software/hardware-based acoustic simulation. Makoto Otani (Graduate School of Eng., Kyoto Univ., Kyoto daigaku-Katsura, Nishikyo-ku, Kyoto 615-8540, Japan, otani@archi.kyoto-u.ac.jp), Takao Tsuchiya (Faculty of Sci. and Technol., Doshisha Univ., Kyotanabe, Japan), Yukio Iwaya (Faculty of Eng., Tohoku Gakuin Univ., Tagajo, Japan), and Yasushi Inoguchi (Ctr. for Information Sci., Japan Adv. Inst. of Sci. and Technol., Nomi, Japan)

The performance of sound field reproduction based on integral equations is restricted by interval of boundary surface discretization, which would cause spatial aliasing when it is not sufficiently fine. Physically accurate reproduction without spatial aliasing throughout an entire audible frequency range would require impractically large number of discretization points. However, the requirement would be relaxed if perceptually satisfactory reproduction is possible even with spatial aliasing. In this presentation, first, we introduce results of perceptual evaluation of effects of spatial aliasing in the sound field reproduction on auditory perception in order to find out a reasonable requirement for realization of perceptually satisfactory system. Second, we introduce development of sound field rendering system consisting of the sound field reproduction system and FDTD (Finite Difference Time Domain) method as an acoustic simulation solver. Finally, we also introduce our attempt to develop a real-time sound field rendering system that utilizes hardware-based acoustic simulation using DHM (Digital Huygens Model) implemented in ASIC (Application Specific Integrated Circuit).

2:20

4pEA5. Sound field reproduction system with active reverberation. Akira Omoto (Faculty of Design, Kyushu Univ., Shiobaru 4-9-1, Minami-ku, Fukuoka 815-8540, Japan, omoto@design.kyushu-u.ac.jp)

Sound field reproduction system can be categorized roughly into two types. The first is physically assured system which reproduce the wavefront or another physical parameters by engineering means. Typical examples are the boundary surface control system, the wave field synthesis, the ambisonics, and so on. The second type is a psychological system that reproduces phantom images one sound at any desired position by artificial manipulation with rather simple technique such as volume based amplitude panning. Examples are the two channel stereo system and the surround system using 5.1 or more loudspeakers. The latter can be recognized as somewhat "artistic" method, because it requires norms based on artistic sense. The present study aims to fuse the engineering and artistic systems mentioned above. The boundary surface control system, which consists of 24-channel narrow directional microphones and 24 channel loudspeakers, was used as the platform of research. Based on the conventional control technique using inverse filtering, additional manipulation such as simple reproduction scheme for high frequency was introduced. Also, the concept of variable reflection acoustic wall system, which adds the artificial reverberation to the sounds that listener makes, was involved in the reproduction system for the direction of the presence.

2:40

4pEA6. Directivity control of a graphene loudspeaker array using elementwise switching technique. Dong-Hwan Kim and Jung-Woo Choi (School of Elec. Eng., Korea Adv. Inst. of Sci. and Technol., 291, Daehak-ro, Yuseong-gu, Daejeon 34141, South Korea, dh.kim92@kaist.ac.kr)

Graphene is a promising material for thermoacoustic transducers because of its low heat capacity per unit area and easy fabrication process. In contrast to the conventional voice-coil loudspeaker, a small thermoacoustic loudspeaker can produce omnidirectional radiation even without enclosures. In addition, the sound level from a thermoacoustic loudspeaker is less sensitive to its radiating surface area, which makes it an ideal omnidirectional sound source in high frequency region. In this work, we present an elementwise switching control of a graphene loudspeaker array utilizing the advantages of thermoacoustic loudspeakers. With an array of 3D graphene transducers, a directivity is controlled by polarity switching of each element according to a pre-determined pattern. A single channel input signal is modulated into two different channel signals using a polarity inverter. Each graphene element is driven by the same signal with different sign, and the polarization pattern of the array forms a designated beam pattern. The proposed technique simplifies the control of beam patterns without using large number of independent channels, and hence, enables the construction of a high frequency driver with variable-directivity.

3:00–3:15 Break

3:15

4pEA7. Cancellation, reproduction, and cloaking using sound field control. Jordan Cheer (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Highfield, Southampton, Hampshire SO17 2LG, United Kingdom, j.cheer@soton.ac.uk)

The active control of sound fields has been applied in active noise cancellation, sound field reproduction, and active acoustic cloaking; however, relatively few studies have considered the relationship between these three areas of research. This paper investigates the physical limitations on active noise control, sound field reproduction, and active acoustic cloaking in terms of the controllability and observability of the sound field. This has been achieved through a series of simulations that consider the control of the sound field in the presence of a scattering object. In terms of the controllability problem, it is shown that while active noise control and sound field reproduction are most effective when considering the interior control problem, active acoustic cloaking is most effective when facing the exterior control problem. Different sensing strategies are then considered for the cancellation, reproduction, and cloaking problems and their impact on the observability of the control problem is discussed.

3:35

4pEA8. Parallel feedback architecture for ambisonics based active noise control. Anshuman Ganguly and Issa Panahi (Elec. Eng., The Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, axg124330@utdallas.edu)

Active noise control(ANC) has been widely studied for smaller acoustic cavities, like cars and small enclosures, over past several years. One of the key challenges in implementing ANC systems for larger acoustic environments lies in extending the target “quiet” zone for maximizing the noise cancellation. Traditional ANC systems use two “omnidirectional” microphones to access reference noise signal and “residual” error signal at target zone. The target zone around the error microphone remains to be small. Feedback ANC with Ambisonics is aimed to extend the quiet zone by employing specialized encoding and decoding of the noise signal accessed using a single Soundfield microphone. In this paper, a parallel Feedback ANC architecture is fused with the Ambisonics encoding to improve the noise cancellation performance. The proposed ANC system also features an adaptive signal decomposition technique to effectively handle noise types with dominant predictable components, like machinery noise. Noise attenuation level (NAL) is used to compare and quantify the advantage of the proposed method. In this paper, an attempt has been made to explore an improved ANC system for real-time noise control in larger acoustic enclosure such as museums and concert halls.

3:55

4pEA9. Open loop active control of noise through open windows. Chuang Shi, Tatsuya Murao, Dongyuan Shi, Bhan Lam, and Woon-Seng Gan (DSP Lab., School of Elec. & Electron. Eng., Nanyang Technolog. Univ., 50 Nanyang Ave., S2-B4a-03, Singapore 639798, Singapore, shichuang@ntu.edu.sg)

Noise propagating through open windows is a big concern of liveability in metropolises, where residential buildings and main roads are built densely. Active noise control (ANC) is a convincing technique to improve the current situation. Classic ANC systems adopt the close loop control. The quietness after control is continuously monitored by error microphones. However, ANC systems that attenuate noise propagating through open windows should ideally avoid the use of error microphones for the ease of implementation and maintenance. Furthermore, as open windows are commonly large as compared to the wavelength of noise, multichannel ANC systems are necessary to be carried out. They result in huge computational burdens that cannot be handled by low-cost hardware platforms. This paper investigates acoustic characteristics of open windows and formulates the control strategies as a spatial sampling and reconstruction problem. This theoretical development prompts the fact that controlling noise propagating through open windows essentially leads to global noise reduction. Both centralized and decentralized open loop control are feasible to deal with broadband noise. The upper bound of the effective band is limited by the spacing between channels, while the lower bound is dependent on the frequency response of secondary sources.

4:15

4pEA10. Virtual sensing technique for feedforward active noise control. Shoma Edamoto (Kansai Univ., 3-3-35, Yamate-cho, Suita, Osaka 5648680, Japan, umaumasound@gmail.com), Chuang Shi (Nanyang Technolog. Univ., Singapore, Singapore), and Yoshinobu Kajikawa (Kansai Univ., Osaka, Japan)

Active noise control (ANC) is one of techniques to reduce unwanted acoustic noise and is based on the superposition property of acoustic waves. If an anti-noise wave is exactly generated to have the same amplitude and inverse phase of the unwanted acoustic noise wave, ANC can reduce the noise level at the desired location, where an error microphone is placed to monitor the error signal. In the case when the error microphone cannot be placed at the desired noise reduction location, virtual sensing techniques are useful. In this paper, we examine one of the virtual sensing techniques in an experiment setup using the parametric array loudspeaker as the secondary source. Experiment results show that improved noise reduction can be obtained at the virtual microphone location.

Session 4pEDa

Education in Acoustics and Signal Processing in Acoustics: Using Computers for Acoustics Education

Hiroshi Suda, Cochair
Dept. of Information and Network Science, Chiba Institute of Technology, 2-17-1, Tsudanuma, Narashino 275-0016, Japan

Victor Sparrow, Cochair
Grad. Program in Acoustics, Penn State, 201 Applied Science Bldg., University Park, PA 16802

Chair's Introduction—1:00

Invited Papers

1:05

4pEDa1. Physics girl experience—Using YouTube to teach acoustics and wave physics. Dianna L. Cowern (PBS, PO Box 9281, San Diego, CA 92169, dianna@physicsgirl.org)

YouTube is usually considered an entertainment medium to watch cats, gaming, and music videos. But educational channels have been gaining momentum on the platform, some garnering millions of subscribers and billions of views. The Physics Girl YouTube channel has focused solely on physics education. Using this channel, we can examine the ways that educational channels grow, their demographical reach, how they are utilized, and how online media and technology can facilitate good and bad learning. YouTube provides unique access to a wide audience. The videos can vary in length and purpose—from tutorials to classroom supplements. In addition, the online world provides the ability to organize and serve educational communities, like PBS Learning Media. With all these tools available, Physics Girl aims to introduce physics as related to real-world examples and interesting questions that are not always covered in traditional teaching. While the goals of each channel vary, each aims to contribute to better STEM learning.

1:25

4pEDa2. Interactive simulators for acoustic education. Hiroshi Suda (Dept. of Information and Network Sci., Chiba Inst. of Technol., 2-17-1, Tsudanuma, Narashino 275-0016, Japan, suda@net.it-chiba.ac.jp)

In the field of acoustic education, there are many complex theories and invisible phenomena. In the lecture of acoustics, the lecturer places much value on forming and expanding mathematical expressions, using textbooks and the blackboard. Students often have difficulty and want new teaching/learning methods and materials. New learning materials can help a student understand a phenomenon; for example, with a short movie describing an actual phenomenon or an environment of experiments using limited models. Now, the calculating power of computers, tablets and smartphones has enough to easily simulate acoustic phenomena. Here distributed web-based interactive simulators for acoustic education are presented.

1:45

4pEDa3. Use of the student edition of ACTRAN in acoustics education. Victor Sparrow (Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu), Jean-Louis Migeot, and Jean-Pierre Coyette (Free Field Technologies, Mont-Saint-Guibert, Belgium)

The power of computation has changed the way problems are solved in acoustics, and students should be trained to have an adequate background in modern techniques and software packages to simulate noise and vibration. Universities often cannot provide industrial-grade packages for their students due to the cost limitations. But recently a student version of a modern acoustics computation package was released, Student Edition of ACTRAN. This tool lets students solve a wide variety of problems using finite elements and infinite elements. This talk will describe the software and how it was used in a graduate level course in computational acoustics. Two examples that will be shown include finding the modes of an irregularly shaped room and computing the acoustic radiation from a gearbox. But it is clear that this software can be used in many different ways, for both undergraduate and graduate acoustics education. Instructors can demonstrate acoustical phenomena for students not adept at using the software themselves, easily visualizing the physics involved in acoustics with the Student Edition of ACTRAN. The software is free for downloading from http://www.mscsoftware.com/page/actran-student-edition.

2:05

4pEDa4. An interactive speech synthesizer for acoustic education developed with pure data. Naofumi Aoki (Graduate School of Information Sci. and Technol., Hokkaido Univ., N14 W9, Kita-ku, Sapporo 060-0814, Japan, aoki@ime.ist.hokudai.ac.jp)

Developing real-time sound systems that allow interactive controls requires advanced skills of sound programming. It often takes time even for experienced programmers to develop such systems. This paper shows an approach for rapid prototyping of such systems by using Pure Data. Especially, this paper focuses on the techniques of interactive controls by combining personal computers and some other peripheral devices such as a smart phone with Touch OSC. This paper describes its possibilities through showing demonstrations. As an example, this paper shows an interactive speech synthesizer. It allows users to control formants by pointing the F1-F2 diagram on the touch screen of a smart phone. Since this system allows time varying controls of the formants, it can synthesize semi-vowel sound as well as vowel sound. In addition, it also allows users to synthesize some other phrases by controlling appropriate movement of the formants by pointing the F1-F2 diagram. These experiences may potentially be effective for students who learn acoustics to be speech therapists. With this system, students can understand the mechanism of speech production in an experimental way. This paper mentions its availability for acoustics education.

2:25–2:40 Break

Contributed Papers

2:40

4pEDa5. Computing the curriculum—Embedding computation in undergraduate acoustics. Matthew C. Wright (ISVR, Univ. of Southampton, Southampton SO17 1BJ, United Kingdom, mcmw@isvr.soton.ac.uk)

I will describe an approach to teaching acoustics to undergraduate engineering students that integrates mathematics, coding, numerical computation, and physical measurement. This approach developed and continues to develop according to my experience as Director of taught acoustical programmes at the Institute of Sound and Vibration Research (ISVR) in the Faculty of Engineering and the Environment at the University of Southampton, UK. Our program is somewhat unusual in that it teaches acoustics in depth to undergraduate students, whose understanding of the physics must develop in parallel with their proficiency in mathematical analysis and coding. I will explore how both numerical computation and coding can be integrated into an acoustics curriculum from the very start, how their inclusion can enhance learning, and how the curriculum can be restructured in order to make best use of these elements, highlighting the features of acoustics as a subject that make it particularly suited to this approach. I will also discuss how the choice of programming languages and environments, and modelling software affect the curriculum and vice versa. I will present examples from my teaching using Jupyter (formerly IPython) notebooks and discuss how to use these effectively.

2:55

4pEDa6. Computer-based acoustics projects for audiology and speech-language pathology students. Jennifer Lentz (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, jjlentz@indiana.edu)

Graduate and undergraduate students studying audiology and speech-language pathology are often required to have a basic working knowledge of acoustics. However, convincing these students the necessity of this knowledge is a common challenge for the acoustics instructors. Consequently, motivating students to be enthusiastic about the material may be facilitated by including methods that replace or supplement traditional teaching methodologies. Computer-based projects that reinforce acoustic concepts learned in the classroom provide an outstanding hands-on opportunity to expose students to the principles of acoustics. Importantly, they also offer clear examples of how knowledge of acoustics is core to the fields of audiology and speech-language pathology. Here, I will illustrate a number of Matlab-based projects that apply acoustics to the topics of hearing, hearing loss, speech production, and speech disorders. These projects are designed to be discipline-specific and also target specifically the abilities and needs of students in audiology and speech-language pathology.

3:10

4pEDa7. Introducing programming through acoustics and audio at Belmont University. Eric W. Tarr (Audio Eng. Technol., Belmont Univ., 1900 Belmont Blvd., Nashville, TN 37212, eric.tarr@belmont.edu)

A course in computer programming has been offered within the department of Audio Engineering Technology at Belmont University to provide an introduction to audio and physics students interested in learning about programming. This course is based on a foundation of acoustics and audio concepts, without requiring previous programming experience. The pedagogical approach has been to expose students to the application of programming within their discipline of interest as a motivation for further exploration. Topics in the course introduce programming concepts for specific signal processing methods. Programming concepts include: variables, operators, data types, as well as basic control structures like loops, conditional statements and functions. While learning programming concepts, students perform various signal processing methods such as: changing a signal's amplitude, signal synthesis, stereo panning functions, mid-side processing, amplitude fades, amplitude modulation, soft/hard clipping, rectification, bit reduction, echo effects, convolution, algorithmic reverberation, expander/compressor dynamic-range processing, and various spectral filters. Course lessons for a "flipped classroom" are freely and publically available at http://www.hackaudio.com/. Since the initial offering of this course, the number of students in the audio engineering program who double-major or minor in computer science has quadrupled.

3:25

4pEDa8. Teaching Mathematica® based computer computations for support of laboratory experiments, special projects, and research for undergraduates taking acoustics and for mentorship students. Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

Undergraduate senior level physics majors taking Acoustics and oceanography majors taking Underwater Acoustics and Sonar learn how to use Mathematica® for doing standard laboratory experiments, special projects, solving homework problems requiring detailed computations or graphics. Students in electrical engineering, economics, mechanical engineering and ocean engineering (along with physics majors) have enrolled in a one credit course—Mathematica® 10 for Scientists and Techies. The feedback from students using a computer language in their course work for the first time is very encouraging and the teaching experience quite rewarding. High school mentorships volunteering to do research after school use Mathematica® for doing projects in the Physical Acoustics Laboratory. Some undergraduate lab projects involve Fourier analysis, measuring sound speed vs. temperature in water, and plotting theoretical and experimental Bessel mode shapes (or cosine shapes) for standing wave resonance in a cylindrical cavity. Here, a long slender tube mounted on the microphone probe translates along the radial or axial directions. Students enjoy generating their own data on a spectrum analyzer, transferring files to Mathematica® and plotting tuning curves vs. frequency in Helmholtz resonators. One student got involved in nonlinear drum vibration while another became interested in synthetic aperture imaging, for senior research projects.

THURSDAY AFTERNOON, 1 DECEMBER 2016 SOUTH PACIFIC 4, 4:05 P.M. TO 5:00 P.M.

Session 4pEDb

Education in Acoustics: Acoustics Education Prize Lecture

Robin Samlan, Chair
Speech, Language, & Hearing Sciences, University of Arizona, P.O. Box 210071, Tucson, AZ 85721

Chair's Introduction—4:05

Invited Paper

4:10

4pEDb1. The role of artificial speech in understanding the acoustic characteristics of spoken communication. Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

Models have long been used to understand the relation of anatomical structure and articulatory movement to the acoustics and perception of speech. Realized as speech synthesizers or artificial talkers, such models simplify and emulate the speech production system. One type of simplification is to view speech production as a set of simultaneously imposed modulations of the airway system. Specifically, the vibratory motion of the vocal folds modulates the glottal airspace, while slower movements of the tongue, jaw, lips, and velum modulate the shape of the pharyngeal and oral cavities, and coupling to the nasal system. The precise timing of these modulations produces an acoustic wave from which listeners extract phonetic and talker-specific information. The first aim of the presentation will be to review two historical models of speech production that exemplify a system in which structure is modulated with movement to produce intelligible speech. The second aim is to describe theoretical aspects of a computational model that allows for simulation of speech based on precise spatio-temporal modulations of an airway structure. The result is a type of artificial talker that can be used to study various aspects of how sound is generated by a speaker and perceived by a listener.

Session 4pMUa

Musical Acoustics: Piano Acoustics and Playing Piano II

Isoharu Nishiguchi, Cochair
Information Media, Kanagawa Institute of Technology, 1030, Shimo-ogino, Atsugi 243-0292, Japan

Nicholas Giordano, Cochair
Physics, Auburn University, College of Sciences and Mathematics, Auburn, AL 36849

Invited Papers

1:00

4pMUa1. Microphone array measurements of concert grand piano soundboards in different stages of production. Niko Plath, Florian Pfeifle, Christian Koehn, and Rolf Bader (Inst. of Systematic Musicology, Univ. of Hamburg, Germany, Neue Rabenstr. 13, Hamburg, Hamburg 20354, Germany, niko.plath@uni-hamburg.de)

A series of measurements is taken on two concert grand pianos in seven different stages of production, starting with the glue-laminated soundboard planks and ending with the completely assembled piano in concert tuned state. Due to the large size of the soundboard as well as its irregular shape, measuring deflection shapes is a nontrivial task. Common measurement tools such as piezoelectric accelerometers can affect the acoustic vibrations of the soundboard due to the added mass. To this end, a noninvasive microphone array method is utilized for the present work. The array consists of 105 microphones successively placed parallel to the soundboard, resulting in a total number of 1289 microphones covering the entire surface. The Soundboard is excited using an acoustic vibrator at 15 positions associated with string termination points on the bass and main bridge. Impulse responses are obtained using the SineSweep technique. The measured sound pressure can be back-propagated to the radiating soundboard surface using a minimum energy method. Based on the measured vibrational and acoustical data a set of signal features is derived which cover direct physical behavior (e.g., driving point mobility, damping, radiation efficiency) as well as perceptive parameters (e.g. attack time, spectral centroid). The empirical findings will contribute to a software tool (based on a real time physical model) to help piano makers estimate the impact of design changes on the generated sound.

1:20

4pMUa2. Analysis of piano performance using an eigenperformance. Masanobu Miura (Dept. of Media Informatics, Ryukoku Univ., 1-5, Seta, Oe-cho, Yokotani, Otsu 5202194, Japan, miura@mail.ryukoku.ac.jp)

Analysis of piano performance has been a great task for musicologists, in particular the "artistic deviation" on performance has been intensively discussed. Introduced here is a method to clarify what occurs on music performance by observing the eigenvector of obtained several piano playings, "Eigenperformance." The eigenperformance is obtained for MIDI-piano playing from the tendency of onset, MIDI-velocity, duration, and instantaneous tempo curves. Eigenperformance is possible to indicate tendencies for a set of piano playing. By using eigenperformance, here investigates 1) the differences in evaluation criteria between pianists and non-pianists, and 2) the differences of deviations due to player's skill. For 1), results of simple scale performance show that pianists evaluate short and sudden tempo changes with taking long time in ending notes along with dynamic and temporal expressions based on phrases on sheet music as good, whereas non-pianists never provide such evaluation. For 2), piano etude "For Elise" is dealt with to be investigated by four classes of proficiency. The eigenperformance revealed that the proficient performance shows same patterns of dynamics among first and second sections in sheet music, whereas the novice performance does not show any analogous on corresponding parts in sheet music.

1:40

4pMUa3. Visualization of Kansei causality for the digital piano sound—Kansei modeling for musical instruments. Rento Tanase and Hideki Sakanashi (Res. & Development Div., Yamaha Corp., 203 Matsunokijima, Iwata, Shizuoka 4380192, Japan, rento.tanase@music.yamaha.com)

Kansei-causality-model (structural causality model of subjective impression) for the digital piano sound and some considerations are presented. At first, the directed graph of hypothetical causality is figured from the report about the performance tests of musical instruments and the interview with the sound engineers. Second, the quantitative Kansei-data acquired by subjective test with semantic-differential method is associated with the directed graph. Then, the Kansei-causality-model are estimated from the directed graph and Kansei-data by SEM (Structural Equation Modeling) / Covariance Structural Analysis. The result suggests structurally the meanings of the words (for example "hiki-gotae (japanese)" and "high grade feeling) for the evaluation of the digital piano sound. The different causality flow represented as two passes on the structural causality model suggests the polysemy of "Noisy. Moreover, the result of CFA (Check Factor Analysis) for the same data indicates the two factors, such as must-be quality and attractive quality, in the sound engineering criterion for the digital piano design.

4p THU. PM

Contributed Papers

2:00

4pMUa4. The role of real-time acquisition of visual and auditory information in the performance of scale and arpeggio tasks in pianists. Chie Ohsawa (Faculty of Music, Kyoto City Univ. of Arts, 13-6 Kutsukake-Cho, Oe, Nishikyo-Ku, Kyoto 610-1197, Japan, ohsawachie@gmail.com), Kenichi Sawai (Graduate Schools for Law and Politics, Univ. of Tokyo, Tokyo, Japan), and Minoru Tsuzaki (Faculty of Music, Kyoto City Univ. of Arts, Kyoto, Japan)

We previously found that the spatial memory of piano keyboard was not accurate enough to play without external spatial cues even in trained pianists. Therefore, the real-time acquisition of spatial information should be essential for piano performance. The aim of the present study was to test how and when the online acquisition of the visual and auditory information contribute to the performance of pianists. Seven undergraduate and graduate students majoring piano performance at a university participated in our experiment. They were asked to play the task sequences, which consisted of scale, arpeggio of triads, alternating fourth and fifth, and alternating fourth, fifth and one octave leap, on an electric piano. Each pianist played in four conditions: “Ordinary,” “No-sound,” “No-vision,” and “No-sound & No-vision.” The locations of the pressed keys were assigned to the locations of the keys to be pressed with dynamic programming after the midi note numbers were converted to the “location numbers.” We analysed the error size and the alignment costs. The results indicated that the visual information clearly contributed for touching accurate place in arpeggio tasks, while the contribution of the auditory information was limited in any task type.

2:15

4pMUa5. Acoustical properties of the front-duplex system in a grand piano. Florian Pfeifle (Univ. of Hamburg, Neue Rabenstrasse 13, Hamburg 20354, Germany, Florian.Pfeifle@uni-hamburg.de)

The duplex scala, patented by Theodore Steinway in 1872, has been subject to several disputes since it found its first application in grand pianos. Not without good reason because, as was shown in the work of Öberg & Askenfeldt (2012), the physical effects explained in the patent can not be substantiated by measurements mainly because the tune of the duplex strings deviate considerably from the suggested theoretical values. Nonetheless, as published in the same work, the perceptual relevance of the rear- and more so of the front-duplex is stressed. As is shown there, even untrained musicians are able to perceive the influence of both duplex scalae. In this treatise, physical properties of front-duplex strings are considered and an explanation of the acoustical importance regarding the string vibration of the main string is given. To this end, a series of high-speed camera measurements of several duplex strings are presented. Due to the small deflection of front-duplex strings, the recorded data is processed using motion magnification methods showing the influence of different modes of vibration in different frequency bands. A physical model of a grand piano string chorus is proposed, aiding in characterizing the main influence of the front duplex string.

THURSDAY AFTERNOON, 1 DECEMBER 2016 HIBISCUS, 3:00 P.M. TO 4:30 P.M.

Session 4pMUb

Musical Acoustics: Musical Instruments from Islands Around the World

Peter L. Hoekje, Chair
Physics and Astronomy, Baldwin Wallace University, 275 Eastland Rd., Berea, OH 44017

Invited Papers

3:00

4pMUb1. Voices of Hawai’i’s rhythmic world. Marques H. Marzan (Cultural Resources, Bernice Pauahi Bishop Museum, 1525 Bernice St., Honolulu, HI 96817, marques@bishopmuseum.org)

Hawai’i is home to a myriad of musical instruments made from a variety of materials, which include stone, wood, gourd, and shell. The resonance that each produces, complements the transmission of knowledge within the Hawaiian culture, through storytelling, dance, and song. Both the secular and spiritual worlds harness the energy these instruments produce, infusing ceremony and daily life with the voices of the uncountable manifestations of akua, the gods. The material choices used in the creation of Hawaiian instruments connect the realm of man to the realm of the gods, as all animate and inanimate materials represent a kinolau, the physical manifestation of the pantheon of Hawaiian gods. The relationships that are forged with our surroundings, through these tools, allow us a glimpse into a world of endless possibilities.

3:20

4pMUb2. Sado Island: Student cultural activities—The drum performance. Shuji Morishita (Music Education, Niigata Univ., 8050, Ikarashi-2-no-cho, Nishi-ku, Niigata, Niigata 9502181, Japan, morishta@ed.niigata-u.ac.jp)

Sado is an island in the Sea of Japan. The population of the island is 57,114 (October 1, 2015). Sado had a gold mine and has a rich natural environment. It was also rich economically and many people came to Sado from many other places in Japan. People brought many entertainment styles to Sado. The traditional arts are still strong, such as Noh drama and Oni-Daiko (ogredrum performances). We have been active in Oni-Daiko performances in Toyooka village since 2009. The festival in Toyooka is held on the first or second Sunday of April every year. It begins in the Shrine of the enshrined deity of the village. We go around the houses in the village and come back to the shrine. It is performed in front of shrines and houses. The students perform it in public at the end of their one-week lesson. We perform it from 7 a.m. to 8 p.m. with the village people. Students mature after taking part in these activities. As the population of Sado begins to decline, we do not know how long we will be able to continue this activity. However, we want to continue carrying out this activity as long as possible.

3:40

4pMUb3. Managing matagia: Instruments and aesthetics in contemporary Tokelau performance. Candice E. Steiner (Univ. of Hawaiʻi at Mānoa, 1890 East-West Rd., Moore 212, Honolulu, HI 96822, candice.steiner@gmail.com)

The hallmark of contemporary Tokelau performances is a significant building of energy over the course of a song and dance, which is driven by gradual tempo increases, upward key modulations, increases in volume, intensification of dance movements, and oneness among performers. The powerful pōkihi (box drum) and accompanying instruments lead the entire group through this unified move from subdued to exhilarating delivery, and when the technical skills of the performers and the cultural meanings of the performance combine in a way that fully engages the performers and audience alike, the resulting energy is palpable. In this presentation, I describe the key musical instruments used in contemporary Tokelau performances, summarize previous research on the instruments' history and use, and discuss the role of the instruments in upholding the Tokelau aesthetics described above.

Contributed Papers

4:00

4pMUb4. Construction and modernization of *Kundun* Pentatonic xylophone. Tobi E. Kemewerigha (Music, College of Education Akakpa, College of Education Akakpa Cross-River, Calabar State, Nigeria, Akakpa CrossRiver, Akakpa 234, Nigeria, kemewerighatobi@yahoo.com) and Stephen G. Onwubiko (Music, Univ. Of Nigeria, Enugu, Nsukka, Nigeria)

This thesis is all about the constructional techniques and modernization of the resonant qualities of Kundun musical instrument from Birom in Plateau State of Nigeria. A discussion of the traditional Xylophone based on the field work conducted in Birom ethnic group understanding a kundun instrument maker, was made. The structure of this Xylophone highlights both the tone qualities and also the use of the cow-horn as pipes which amplifies the resonance of the slab. The construction and methods used here embodied traditional technology together with modern innovations, with the aim of improving the tone and aesthetic qualities of the instrument. This report also offered guides on the correct choice of wood and other materials for construction. Special attention was given to the acoustic behavior of kundun and the particular problems of the tropical climate conditions, to which the instrument could be exposed and to the overall tone quality of the instrument during performance. Some scientific approaches, in terms of the accurate measurements and amplification of sound, were taken into cognizance so as to enhance the knowledge of the construction of kundun musical instruments and its aesthetic quality.

4:15

4pMUb5. Calculating scale tunings of Polynesian nose flutes. Peter L. Hoekje (Phys. and Astronomy, Baldwin Wallace Univ., 275 Eastland Rd., Berea, OH 44017, phoekje@bw.edu)

The Polynesian nose flute is found in two versions that are blown in a similar manner but fingered and tuned differently. The Eastern Polynesian flute, known in Hawai'i as *'ohe hano ihu,* but is found also to Tahiti and Niue. The tube is closed by a segment node at one end but with a hole for blowing and open at the other end. It typically has two or three finger holes along the tube and three or four playable notes. Being similar to other flutes, the scale tunings can be calculated from hole and tube dimensions by analytic expressions. In Fiji, New Zealand, Samoa, and Tonga, where it is known as *fangufangu*, both ends of the Western Polynesian flute are closed by segment nodes. There are five holes are placed along the top, usually evenly spaced from one end to the other, with a sixth hole in the middle underneath, giving a variety of fingering options with four to six notes and two registers. The scale tunings are determined by first calculating the frequency response using a lumped-element method.

THURSDAY AFTERNOON, 1 DECEMBER 2016 HIBISCUS, 4:30 P.M. TO 5:15 P.M.

Session 4pMUc

Musical Acoustics: Musical Instruments from Islands around the World II—A Concert

Peter L. Hoekje, Chair
Physics and Astronomy, Baldwin Wallace University, 275 Eastland Rd., Berea, OH 44017

In Hawai'i, hula complements Hawaiian storytelling presented in song form. Various instruments are also used in accompaniment of these songs, most often by ipu, gourd drums, pahu, wooden drums, and kāla'au, dance sticks, either used by the dancer or the chanter. Various instruments help to emphasize the stories, through tone and cadence.

THURSDAY AFTERNOON, 1 DECEMBER 2016 SOUTH PACIFIC 3, 1:00 P.M. TO 2:15 P.M.

Session 4pNSa

Noise: Prediction of Transportation Noises and Other Topics II

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

Shinichi Sakamoto, Cochair
5th Department, Institute of Industrial Science, The University of Tokyo, 4-6-1, Komaba, Meguro-ku, Tokyo 153-8505, Japan

Contributed Papers

1:00

4pNSa1. Experimental contribution analysis of external noise components to interior noise of automobile. Seongil Hwang (Mech. Eng., Texas A&M Univ., 3123 TAMU, College Station, TX 77843), Myunghan Lee, Kang Duck Ih (Hyundai Motor Co., Hyundai Motor Co., Hwaseong, Geongi-do, South Korea), and Yong-Joe Kim (Mech. Eng., Texas A&M Univ., College Station, TX, joekim@tamu.edu)

The contribution analysis of various noise sources to interior noise is important, enabling to design an automobile with a low interior noise level. Here, using a modified Cholesky Decomposition (CD), it is proposed to decompose interior noise spectra into multiple spectra, each represents the contribution of a specific noise source to the interior noise. Through an experiment with the two speakers driven by two independent white noise signals, it is shown that the measured noise spectrum can successfully be decomposed into two contributions, each associated with noise radiated from one of the two speakers. Then, an automobile was tested on a road at the speeds of 65 mph and 80 mph. In this experiment, 64 external and 4 interior microphones were used to measure external noise source signals and interior noise signals, respectively. The contribution results obtained by processing the measured microphone signals show the highest contributions of the aeroacoustic sources on the windows and the hood to the interior noise. It was also observed that the contribution of the windows increased as the speed increased and that the higher contribution to the interior noise at a specific seat could be obtained from the window closer to the seat.

1:15

4pNSa2. Noise produced within spaces adjacent to high performance jet operations on large amphibious assault ships. Ronald Hughes, Steven Antonides, and Chandrasekhar Kannepalli (NSWCCD, 9500 Macarthur Blvd., West Bethesda, MD 20817, ronald.g.hughes@navy.mil)

Airborne noise levels have been measured in spaces adjacent to F-35B operations in proximity to the flight deck on Large Amphibious Assault Class ships (L Class). Noise levels observed in some manned spaces are hearing hazardous. The noise produced during high performance jet operations is intermittent to steady state in duration. Results reported here are only for vertical landings (VL) and do not include short takeoffs (STO); however, noise due to both VL and STO exhibit similar characteristics. Observed VL noise features include two prominent types of noise sources during aircraft decent as well as source directionality. These features are compared to LES CFD predictions for a supersonic jet impinging on a plate. Qualitative features observed in data have been included in a noise model to predict noise levels in Gallery deck compartments. Predicted noise levels are used to ascertain noise treatments and hearing protection on L Class ships during F-35B air operations. Motivations for controlling noise levels are to prevent or reduce hearing threshold shifts in crew members and to promote hearing recovery and communication intelligibility. Noise characteristics of F-35B landings and diagrams of predicted noise levels that address noise-induced hearing loss, hearing recovery, and communication intelligibility are shown.

1:30

4pNSa3. Detached eddy simulation of a highlift airfoil noise with spectral difference method. Junhui Gao and Xiaodong Li (Beihang Univ., New Main Bldg., D401, Beijing 100191, China, gaojhui@buaa.edu.cn)

In this study the noise generated by the 30P30N highlift airfoil is simulated with a high order computational aeroacoustics code. To handle the complex geometry of the airfoil, the high order spectral difference method based on unstructured mesh is used for spatial discretization. The multi-time-step method based on Adam-Bashforth scheme is utilized for time marching, and a speed-up ratio greater than 10 is obtained comparing with the single time step method. The detached eddy simulation method is conducted for turbulence modeling. Radiation and outflow non-reflective boundary conditions are used in the far field and downstream. In this simulation, the inflow Mach number is 0.17, and the Reynolds number based on the inflow velocity and the chord length of the airfoil is 1.7e6. Three attack angles, which are 4, 5.5, and 8.5 degrees, respectively, are considered in this study. The dynamic pressure on the surface of the slat is sampled and compared with the experimental data by other researchers, and a good agreement is obtained. The far field noise, computed with the permeable Ffowcs Williams-Hawkings (FW-H) integration method, is analyzed and compared with the experimental data. The noise source is analyzed and identified with a correlation analysis of the dynamic pressure on the slat and the near field noise.

1:45

4pNSa4. Three-dimensional computational fluid dynamics prediction of mild surge in a turbocharger compression system vs. experiments. Rick Dehner, Ahmet Selamet, and Emel Selamet (Mech. Eng., The Ohio State Univ., 930 Kinnear Rd., Columbus, OH 43212, dehner.10@osu.edu)

The flow instabilities in a turbocharger compression system are studied numerically by employing a compressible, three-dimensional computational fluid dynamics code. The model represents the full compression system of a turbocharger test stand, consisting of a compressor inlet duct breathing from ambient, a centrifugal compressor, an exit duct connected to a plenum, followed by another duct which incorporates a valve restriction. The detailed compressor model includes the full rotating impeller and resolution of the clearance gap between the impeller blades and shroud. The simulation begins at a converged, stable operating point, where the flow rate at the outlet boundary is gradually reduced and maintained at the final value. Characteristics of mild surge are captured as the mass flow rate is reduced below the stability limit, including a discrete low frequency sound peak near the Helmholtz resonance of the compression system. The predictions are then compared with experimental results obtained from the turbocharger stand placed in a hemi-anechoic room. The computational results are shown to be capable of reproducing a number of key experimental observations, including the details of low-frequency pressure fluctuations in the compressor ducts and plenum, and the transition from stable operation to oscillating mild surge.

2:00

4pNSa5. Numerical Simulation of time domain wall impedance on circular and annular ducts using the immersed boundary method. Braulio Gutierrez Pimenta (Mech. Eng. Dept., Univ. of Brasília, Universidade de Brasília, Campus Universitário Darcy Ribeiro, Faculdade de Tecnologia, Departamento de Engenharia Mecânica, Brasília, Distrito Federal 70910-900, Brazil, braulio.pimenta@gmail.com), Ana Luisa P. Maldonado (Rolls-Royce University Technol. Ctr., Univ. of Southampton, Southampton, United Kingdom), and Roberto F. Bobenrieth Miserda (Mech. Eng. Dept., Univ. of Brasília, Brasília, Brazil)

In this work, time domain boundary conditions of wall impedance are simulated on circular and annular ducts of constant cross section. The discretization technique used here is the immersed boundary method in the framework of finite volume formulation. It has fourth order of precision in space and uses a third order Runge-Kutta method for time stepping. Several boundary conditions for wall impedance are tested and then compared with the analytical results of the one-dimensional resonating chamber for code validation. The impedance boundary conditions are simulated on a circular duct with wall treatment and with a synthetic sound source. Mean flow conditions include constant axial velocity and swirling flows of rigid body rotation and potential flow. The results are then compared with a fourth order eigensystem method of linearized governing equations. The results of the immersed boundary methodology are assessed along with the wall impedance boundary condition models and their linear validity is compared with the results of the eigensystem method.

Session 4pNSb

Noise: Low Frequency Noise: Measurement, Evaluation, Enforcement, and Policy

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

Yukio Takahashi, Cochair
National Institute of Occupational Safety and Health, Japan, 6-21-1 Nagao Tama-ku, Kawasaki 214-8585, Japan

Invited Papers

2:30

4pNSb1. The state of low frequency noise regulation in the United States. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1147, les@nonoise.org) and David S. Woolworth (Oxford Acoust., Oxford, MS)

This paper surveys state and local regulation of low frequency noise in the United States. Included in the survey are the noise regulations of the 500 largest cities in the United States, state regulations from the dozen states with reasonably comprehensive noise regulations, and various other regulations, usually related to mineral extraction or energy production. The prevalence of low frequency criteria in noise regulations as well as the metrics and criteria used are discussed and evaluated.

2:50

4pNSb2. Case report of a low-frequency noise complaint settled through adjudication by the Environmental Dispute Coordination Commission. Yukio Takahashi (National Inst. of Occupational Safety and Health, Japan, 6-21-1 Nagao Tama-ku, Kawasaki, Kanagawa 214-8585, Japan, takahay@h.jniosh.johas.go.jp)

In Japan, complaints about environmental low-frequency noise have been increasing in recent 20 years. The Environmental Dispute Coordination Commission is a subsection of the Ministry of Internal Affairs and Communications and one of its main duties is to resolve environmental disputes promptly and appropriately through adjudication, mediation, and other ways. The Commission treats some cases of complaint about low-frequency noise every year. This presentation outlines the activity of the Commission and reports a case of a low-frequency noise complaint settled through adjudication by the Commission. The author took part in the adjudicating process as an external technical expert, and supported on-site measurement of the noise and evaluation of its effect. The source of the noise in the case was air compressors located at a supermarket adjacent to the complainant's house. In evaluating the effect of the noise, much importance was given not to merely the sound pressure level of the noise but to the correspondence between the sound pressure level of the noise and subjective ratings of the complainant's annoyance. Thanks to a good correspondence found between them, the complainant and the supermarket arrived at compromise.

3:10

4pNSb3. Characterization of low frequency sounds emanating from entertainment venues. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

The advent of inexpensive low frequency reproduction for audio has changed the way modern popular music is composed and recorded; these inexpensive systems are more common than ever in clubs (and vehicles), which has a direct effect on the soundscape resulting from high sound level playback and limited sound attenuation for low frequencies in typical building structures. This paper characterizes the low frequency content of sounds propagated intra-building, inter-building, and to the outside from various entertainment venues from the perspective of both frequency and time. Additionally, the paper will discuss suggested techniques for testing venues under controlled situations with pre-recorded and live sound sources to augment testing based on standards.

3:30

4pNSb4. Psychological and physiological response of low frequency noise of ordinary persons and complainants. Shinji Yamada (The Univ. of Yamanashi, 1-9-3 Shiobe, Kofu 400-0026, Japan, shinji-yamada@ae.auone-net.jp), Yukio Inukai, Tsutae Sebayashi (Non Profitable Organization on Noise, Vib. and Low Frequency Noise, Kofu, Japan), and Toshiya Kitamura (The Univ. of Yamanashi, Kofu, Japan)

In Japan, there are some complainants of low frequency noise. There are two sorts of complains on low frequency noise. One complain is the phenomenon of rattling noise. Light windows or doors are vibrated and rattled by the pressure of low frequency noise. Sound sources are a big diesel engine, pass by of a heavy truck on a highway bridge, a gas turbine engine generator etc. The other complaints are the direct hearing or detection of low frequency noise. Sound sources are a boiler, a cooling tower, etc., near neighbors, or solid borne noise inside a room. But recently, there are some complains at very low level and the sound source cannot be detected. In these

cases, there may be much influences by mass media and Internet knowledge. In this paper we explain the mechanism of complains on low frequency noise as bellows. Hearing thresholds of ordinary persons and complainants. Physiological response of ordinary persons and complainants. Time history of annoyance of complainants according to on-and-off of sound source. Recent complains against low frequency noise. Guideline on low frequency noise: Mechanism of complaint in brain to wind turbine low frequency noise.

3:50–4:05 Break

Contributed Papers

4:05

4pNSb5. Numerical analysis of low frequency noise propagation from outdoor into house. Dai Hashimoto (Yamaha Corp., 10-1, Nakazawa-cho, Naka-ku, Hamamatsu, Shizuoka 430-8650, Japan, dai.hashimoto@music.yamaha.com), Naohisa Inoue, and Tetsuya Sakuma (Graduate School of Frontier Sci., Univ. of Tokyo, Kashiwa, Chiba, Japan)

There have been increasing environmental problems caused by low-frequency noise: window rattle, human annoyance and so on. Sound transmission from outdoor into a house in the low-frequency range is a considerably complex phenomenon because the house behaves as a total vibrating system composed of windows, walls, frames, ventilation holes, and so on. This paper presents a finite element model of a simple house in order to investigate the low-frequency sound transmission. The features of the model are as follows: 1) a rigid hollow cube with an aperture on its one side is placed on a rigid plane, 2) a windowpane mounted on the aperture is modeled as a limp membrane or an elastic plate, 3) the outdoor domain is truncated by perfectly matched layers in order to simulate a semi-free field, and 4) a noise source is supposed to be an air conditioning outdoor unit placed near the house. It is confirmed that the indoor SPL gains mainly due to two resonance systems: one is the global motion composed of indoor air tightness and mass around the aperture, and the other is the series of normal modes excited in a cubic room.

4:20

4pNSb6. Influence on human body of audibility amplitude modulation low-frequency noise in residential environment. Hiroshi Matsuda and Nobuo Machida (College of Sci. and Technol., Nihon Univ., 7-24-1 Narashinodai, Funabashi, Chiba 2748501, Japan, matsuda@eme.cst.nihon-u.ac.jp)

Low-frequency noise of about 100 Hz including audible frequency is positioned as one of the factors of environmental problem. First, this paper describes countermeasures of psychological complaint by low-frequency noise generated by residential environment in Japan published in the Ministry of the Environment. Next, we experimentally examined the human body influence of audibility amplitude modulation low frequency noise (AMLFN) on the psychological and physiological response measured by whole-body exposure. The AMLFN used by this research fluctuated the fluctuation cycle of the carrier wave and the sound pressure level periodically by the amplitude modulation. The measuring objects of the psychological response are an annoyance by the psychological questionnaire and an impression evaluation by the semantic differential method. The measuring objects of the physiological response are a salivary amylase activity, a R-R interval, and a power spectrum of heart rate variability. The stress reaction by AMLFN exposure was examined from these measurement results. From these experimental results, the stress reactions during AMLFN exposure were tendency increased with an increase of the modulation depth of the sound pressure level and a decrease of the fluctuation cycle, and the annoyance was tendency increased with a decrease of the carrier frequency.

4:35

4pNSb7. Wind-induced noise in a convective boundary layer. Carl R. Hart and D. K. Wilson (U.S. Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@usace.army.mil)

Wind-induced noise, as sensed by a screened microphone in the atmosphere, produces a highly varying noise floor. At low frequencies, this noise source may dominate the ambient noise spectrum and present a challenge for signal detection. The sensed noise is affected by several factors in the atmospheric convective boundary layer: the mean flow at the height of the microphone, turbulent eddy interaction with the windscreen, and pressure fluctuations within the turbulent flow. Recently, van den Berg (2006) compared measured wind noise to modeled spectra. In that study, several meteorological variables were inferred from fits of modeled wind noise spectra rather than from meteorological observations. If additional meteorological observations are collected, then it may be possible to conduct model validation. In addition to temperature, pressure, and wind velocity, the surface roughness length for momentum and the vertical temperature flux must be observed. Recent models of turbulent wind velocity, in the convective boundary layer, suggest that the planetary boundary layer height may also be an important variable for wind-induced noise. This study analyzes data collected in the Southwestern United States during a 2007 long-range sound propagation experiment. Wind noise data are compared against concurrent meteorological data and several wind-induced noise models.

4:50

4pNSb8. Low frequency sound sources, regulations, and the wind noise evaluation for appropriate procedure against complaints. Masayuki Shimura (INCE/Japan, 2-23-2, Higashi-ikebukuro Toshima-ku, Tokyo 170-0013, Japan, shimura@kensetsukankyo.co.jp)

There are many kinds of low frequency noise sources, including cannon, tunnel-blasts in construction, the Shinkansen, arch dams, elevated roads, wind farms, and so on. For example, low frequency noise by bombardment practice of the Ministry of Defense is dominant in the 8—16 Hz bands in the emission area nearby dwelling houses. Local regulations for soundproofing are applied in such cases. In the sound emission problems by drainage from the dam, we have to discuss the noise abatement of low frequency sound. However, there is neither existing regulation nor environmental assessment. Individual correspondence has been done for each noise source based on the complaints from dwellers. There are not unified countermeasures for low frequency sources in Japan. Evaluation of low frequency noise is very difficult because the measuring results are influenced by wind pressure fluctuation in natural wind. The physiological and psychological influence by such noise is also not clear. In this presentation, a new method to measure low frequency sound in natural wind is proposed.

Session 4pPA

Physical Acoustics: Nonclassical Nonlinear Acoustics of Solids

Lev A. Ostrovsky, Cochair
PSD, NOAA ESRL, 325 Broadway, R/PSD99, Boulder, CO 80305

Paul A. Johnson, Cochair
Geophysics, LANL, MS D443, Los Alamos, NM 87545

Yoshikazu Ohara, Cochair
Department of Materials Processing, Tohoku University, 6-6-02 Aoba, Aramaki-aza, Aoba-ku, Sendai 980-8579, Japan

Invited Papers

1:00

4pPA1. Propagation of a finite-amplitude pulse in a bar of berea sandstone: The mechanisms of classical nonlinearity, conditioning, and hysteresis. Marcel C. Remillieux, James A. Ten Cate, Pierre-Yves Le Bas, and T. J. Ulrich (Los Alamos National Lab., Geophys. Group (EES-17), Mail Stop: D446, Los Alamos, NM 87545, mcr1@lanl.gov)

We study the propagation of a finite-amplitude pulse in a slender bar of Berea sandstone. The center frequency of the pulse and aspect ratio of the bar are such that the problem can be adequately described by the propagation of a longitudinal wave in a 1D system. The evolution of the three Cartesian components of the particle velocity on the surface of the bar as functions of the propagation distance and source amplitude is carefully monitored without contact using a 3D laser Doppler vibrometer. In these experiments, we evidence simultaneously the effects from classical nonlinearity, hysteresis, and conditioning (i.e., elastic softening) in the impulsive waveforms, as the pulse propagates away from the source. Traditionally, this type of experiments has been conducted to quantify only classical nonlinearity, through the parameter β, based on the amplitude growth of the second harmonic as a function of the propagation distance. In this work, we also use these experiments to quantify conditioning, through the parameter of nonclassical nonlinearity α, based on the relative change in the arrival time of the pulse as a function of strain and distance from the source.

1:20

4pPA2. Creep-induced nonlinear ultrasonic change in an austenitic stainless steel. Toshihiro Ohtani, Yutaka Ishii (Mech. Eng., Shoan Inst. of Technol., 1-1-25, Tsujido-nishikaigan, Fujisawa, Kanagawa 2518511, Japan, ohtani@mech.shonan-it.ac.jp), Masaaki Tabuchi, Kota Sawada, and Hiromichi Hongo (Creep Resistant Mater. Group, National Inst. for Mater. Sci., Tsukuba, Ibaraki, Japan)

We investigated the evolutions of two nonlinear acoustic characterizations: resonant frequency shift and three-wave mixing, with electromagnetic acoustic resonance (EMAR) throughout the creep life in an austenitic stainless steel, JIS-SUS 304. EMAR was a combination of the resonant acoustic technique with a non-contact electromagnetic acoustic transducer (EMAT). We used bulk- shear-wave EMAT, which transmits and receives shear wave propagating in thickness direction of a plate specimen. Creep tests with plate specimens were carried out at 973 K, and 100 MPa and interrupted at several time steps. Two nonlinear acoustic parameters showed peas at 40% of creep life. After that they deceased, they increased from 60% of creep life to rupture. We interpreted these phenomena in terms of dislocation recovery, recrystallization, and restructuring and the initiation and growth of creep void, with support from the SEM and TEM observation.

Contributed Papers

1:40

4pPA3. Nonclassical nonlinearity and crystal defects in lithium niobate. Chandrima Chatterjee, Lucien Cremaldi (Phys. and Astronomy, Univ. of MS, Lewis Hall, Rm. 108, University, Oxford, MS 38677), and Igor Ostrovskii (Phys. and Astronomny, Univ. of MS, University, MS, iostrov@phy.olemiss.edu)

Nonclassical nonlinearity (NN) in ultrasound attenuation and acoustic memory in $LiNbO_3$.may be connected to crystal defects. The optical spectra of photoluminescence (PL) and absorption taken from 350—900 nm reveal multiple point defects and impurities such as F-center, Nb_{Li}^+, Fe, Ba, Ar, Xe and others. The spectra are taken at room temperature from the Czochralski grown bulk crystals and wafers. The samples are scanned by taking PL spectra from different locations along the z- and y-axes. The data from several samples reveal a nonuniform distribution of the defects along the crystallographic axes. The dependencies of PL peaks versus position have a sine-like component on top of some constant intensity. For instance, a scan for F-center has a period (P) from 0.27 to 0.75 mm and for Ba impurity from 0.25 to 0.75 mm in bulk crystals. In the wafers, P may be smaller, that is 0.1 mm and smaller. However, a ratio of maxima to minima in PL peaks is higher for bigger P. The overall spread of P is roughly from < 0.1 mm to about 1 mm. Some defects are electrically charged, and may influence the

piezoelectric properties. Consequently, a real crystal has some constants which are nonuniform along its axes. The measured periods P correspond to ultrasonic wavelengths in the Megahertz frequency range. The comparison of the periods P in defects distributions with the frequencies of NN phenomena points to considering crystalline defects as those responsible for the NN observed in $LiNbO_3$.

1:55

4pPA4. Ultrasonic array imaging of contact-acoustic nonlinearity. Jack Potter, Jingwei Chen, Bruce Drinkwater, and Anthony Croxford (Mech. Eng., Univ. of Bristol, Queen's Bldg., University Walk, Bristol BS8 1TR, United Kingdom, jack.potter@bristol.ac.uk)

Classically, ultrasonic arrays achieved physical beam-forming through the application of phase delays to the parallel transmission of elements. Alternatively, beam-forming may be emulated through post-processing of sequentially transmitted array data. Though exactly equivalent for linear systems, these parallel and sequential fields differ in their nonlinear propagation. Information pertaining to elastic nonlinearity is encoded onto the relative parallel/sequential field by interference between the field components of individual element transmissions. A family of nonlinear imaging techniques has been developed which exploit differences between parallel and sequential fields in order to image elastic nonlinearity. Diffuse energy and coherent scattering variants of the technique are demonstrated for the imaging and characterisation of contact-acoustic nonlinearity in application to the detection and sizing of closed fatigue cracks.

2:10

4pPA5. Simulation of coupled structural-acoustic response with dynamic damage evolution. Jonathan S. Pitt (The Penn State Univ., Appl. Res. Lab, PO Box 30, Mailstop 3320B, State College, PA 16804, jonathan.pitt@psu.edu)

A novel time-domain method for simulating dynamic damage evolution in a coupled structural-acoustic system is presented. The system is derived via the theory of continuum damage mechanics, and incorporates standard damage evolution models, but is readily extendible to more exotic formulations. The overall solution method is staggered, solving for the dynamic damage evolution first with an explicit step, and then using the new values in the coupled computation of the structural-acoustic system. The spatial domain is discretized using a mixed finite element method, and the temporal domain is discretized with a higher-order implicit time discretization scheme. Efforts toward fully coupled verification of the solution algorithm are presented, as are validation studies for cases without evolving damage. Applications with evolving damage are presented, and present a first principles study of changes in the structural acoustic response to dynamically evolving damage in the structure. Special attention is given to brittle fracture. Examples of downstream usage of the evolving structural response are discussed in the concluding remarks.

2:25

4pPA6. Evaluation of fatigue damage in a pure copper with nonlinear three-wave interaction. Yutaka Ishii (Graduate School of Eng., Shonan Inst. of Technol., 1-1-25, Tsujido-nishikaigan, Fujisawa, Kanagawa 2518511, Japan, 11t1501@sit.shonan-it.ac.jp), Toshihiro Ohtani (Mech. Eng., Shoan Inst. of Technol., Fujisawa, Kanagawa, Japan), Masayoshi Nakaniwa (Nippon Steel & Sumikin Technol., Amagasaki, Hyogo, Japan), and Masayuki Kamaya (Inst. of Nuclear Safety System, Inc., Mikata-gun, Fukui, Japan)

We investigated nonlinear ultrasonic characterization, nonlinear three-wave interaction, of the pure copper during fatigue with EMAR (Electromagnetic Acoustic Resonance), which was the combination with ultrasonic resonance and non-contacting transducer, EMAT (Electromagnetic Acoustic Transducer). In nonlinear three-wave interaction method, two intersecting ultrasonic waves produced a scattered wave when the resonance condition was satisfied. The amplitude in resonant scattering wave was measured. Nonlinear three-wave interaction method exhibited high sensitivity to micro-structural change of the damaged material. It rapidly increased from 50% of fatigue life to the fracture. TEM (Transmission Electron Microscope) and EBSD (Electron Backscatter Diffraction) observations supported this phenomenon caused by dislocation movement. The sensitivity in three-wave interaction method was higher than that in linear methods. The non-contact resonance-EMAT measurement can monitor the evolution of nonlinearity throughout the fatigue life and has a potential to assess the damage advance and to predict the fatigue life of metals.

2:40

4pPA7. Resonant acoustic nonlinearity of defects: Non-classical manifestations and applications for diagnostic imaging. Igor Solodov (Inst. for Polymer Technol., Univ. of Stuttgart, 32 Pfaffenwaldring, Stuttgart 70569, Germany, igor.solodov@ikt.uni-stuttgart.de)

Acoustic nonlinearity of non-bonded interface (Contact Acoustic Nonlinearity (CAN)) is a type of non-classical nonlinearity perspective for applications in non-destructive evaluation of cracked defects. The presence of such defects also leads to a local decrease in rigidity and manifests in a characteristic frequency of the defect. A frequency match between the driving acoustic wave and this frequency provides a Local Defect Resonance (LDR) and an efficient energy delivery from the wave into the defect. A combination of CAN and LDR, therefore, manifests a profound nonlinearity even at moderate acoustic excitation level. Under resonance conditions, a strong enhancement of the higher harmonic (HH) amplitudes is observed. A high quality factor of LDR is also used as a filter/amplifier/resonator for efficient frequency mixing. These "conventional" nonlinear effects are not the only dynamic scenario of nonlinear phenomena for resonant defects. A combined effect of LDR and CAN also results in qualitatively new features characteristic of nonlinear and parametric resonances. Due to parametric instability, nearly total input energy at fundamental frequency can be converted into HH or subharmonic vibrations of the defects. Both super- and subharmonic LDR are strongly localized in the defect area that provides a background for highly-sensitive defect-selective imaging.

2:55

4pPA8. In situ nonlinear ultrasonic probing during plastic deformation. Carolina Espinoza, Vicente Salinas, Nicolás Mujica, and Fernando Lund (Phys., Universidad de Chile, Av. Blanco Encalada 2008, Santiago 837.0415, Chile, deplasticoverde@gmail.com)

In is well known that nonlinear ultrasound is sensitive to micro-structural changes in materials. Here, we present recent results obtained during tensile tests of aluminum samples. Using continuous ultrasonic waves, we study the second harmonic generation and its relationship with dislocation proliferation during plastic deformation. The experimental setup is based on two transducers, an emitter and a receiver, both coupled face to face to a probe mounted on a standard tensile test machine. We measure both the first and second harmonic received amplitudes, A_{ω} and $A_{2\omega}$ respectively, for several amplitudes of the emitted signal. These are related by a nonlinear parameter β, through $A_{2\omega} = \beta A_{\omega}^2$. We observe that β is independent of the applied stress during the elastic regime, whereas it presents strong variations once plasticity occurs, presenting a pronounced maximum before fracture. This maximum can be considered as a signature, a warning, of a fracture that is soon to come.

3:10–3:25 Break

Invited Papers

3:25

4pPA9. Fast and slow nonlinear elastic response of disparate rocks and the influence of moisture. Jacques Riviere (GeoSci., Penn State Univ., Deike 510, University Park, PA, jvr5626@psu.edu), Parisa Shokouhi (Civil Eng., Penn State Univ., University Park, PA), Robert A. Guyer, and Paul A. Johnson (EES-17, Los Alamos National Lab., Los Alamos, NM)

We study nonlinear elastic phenomena in rocks at the laboratory scale, with the goal of characterizing and understanding observations at crustal scales, for instance, during strong ground motion and earthquake slip processes. A dynamic perturbation of microstrain amplitude in rocks results in a transient elastic softening followed by a log(t)-type relaxation back to the initial unperturbed elastic modulus as soon as the excitation is removed. Here we use Dynamic Acousto-Elastic Testing (DAET) to investigate the relaxation behavior over 7 orders of magnitude in time (from 10^{-4}s to more than 10^{3}s). We find that relaxation starts for all samples between 10^{-3} and 10^{-2}s. Some samples then exhibit a nearly perfect log(t)-relaxation, implying that no characteristic time can be extracted, while some other samples show a preferential relaxation around 0.1s/1s. Such features appear insensitive to the amplitude of the dynamic perturbation and to the moisture content within the sample. The full nonlinear elastic response (fast dynamics) is also extracted at all amplitudes and moisture content. Adsorption of water on the grains strongly increases the elastic softening during the dynamic perturbation and the non-classical nonlinear features, whereas the classical features seem rather unaffected.

3:45

4pPA10. Steady-state analysis of nonlinear ultrasonic waves caused by crack face friction. Taizo Maruyama and Terumi Touhei (Civil Eng., Tokyo Univ. of Sci., 4th Fl., 5th Bldg., Noda Campus, 2641 Yamazaki, Noda City, Chiba Pref. 278-8510, Japan, taizo_maruyama@rs.tus.ac.jp)

In general, the detection of closed cracks is difficult by using the conventional linear ultrasonic testing. Therefore, the nonlinear ultrasonic testing (NLUT) based on the contact acoustic nonlinearity (CAN) is expected to be an effective technique for the problem. Several studies have reported the presence of nonlinear ultrasonic waves (higher- and sub-harmonic waves) due to the CAN. However, the theoretical explanation of the physical phenomena is not sufficient at present. For accurate NLUT, theoretical investigation of the nonlinear vibration phenomena of crack faces is desirable as a basic research. It is true that the time-domain analysis is becoming possible for the purpose. However, it is difficult to investigate in detail steady-state vibration of crack faces, which is supposed to affect behavior of nonlinear ultrasonic waves significantly. In this study, a novel numerical method is developed for the NLUT by coupling the boundary element method (BEM) and harmonic balance method (HBM) in order to investigate the above steady-state vibration. The scattering model is for the 2-D SH wave field that causes the crack face friction. Several numerical examples are presented in order to demonstrate validity and effectiveness of the proposed method.

Contributed Papers

4:05

4pPA11. The soil plate oscillator: Measuring nonlinear mesoscopic elasticity of granular medium in flexural vibration. Emily V. Santos and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

The soil plate oscillator apparatus (SPO) consists of two circular flanges (5 inch ID, 8 inch OD, 2.5 inch thick) sandwiching and clamping a thin (1/8 inch) circular elastic acrylic plate. Granular media (soil, masonry sand, glass beads, or even edible granular material) is supported at the bottom by the plate and side walls of the upper flange. A magnetic disk (1 cm) fastened below the plate (on center) is driven by a swept sinusoidal current in a coil placed below the magnet. This electrodynamic SPO, placed in a Wheatstone bridge, was used to measure the on-axis motional electrical impedance Z_{mot} with and without granular loading. Nonlinear tuning curves near resonance exhibit softening with increasing drive amplitude. Measurements of Real{-Z_{mot}} and Imag{Z_{mot}} were used to predict the nonlinear mechanical point impedance (Z_{mech} = constant / Z_{mot}) behavior of the SPO system. A lumped element model (granular media and elastic plate system) help determine the mesoscopic nonlinear properties of the granular media alone. Nonlinear tuning curves of the magnet's acceleration were also measured. A bilinear hysteresis model fits the tuning curve results. See I. T. Perez-Miravete *et al.*, JASA **125**, 1302 (2009) for a nonlinear mesoscopic analysis of a flexural bar.

4:20

4pPA12. Second harmonic ultrasonic waves detection using a double-layered piezoelectric transducer. Makoto Fukuda and Kazuhiko Imano (Dept. of Mathematical Sci. and Electrical-Electronic-Comput. Eng., Graduate School of Eng. and Resource Sci., Akita Univ., 1-1 Tegata Gakuenmachi, Akita 010-8502, Japan, mfukuda@gipc.akita-u.ac.jp)

A real-time detection system for second harmonic ultrasonic pulse waves using a double-layered piezoelectric transducer (DLPT) has been constructed. We also analyzed the effective detection of the second-harmonic ultrasonic pulse waves generated by closed-cracks and/or contact surfaces of solids (CAN: contact acoustic nonlinearity) using DLPT and the pulse inversion averaging (PIA) method. Lamb waves were also detected in a double-layered piezoelectric transducer (DLPT). The second harmonic components generated from a closed crack in the glass plate were detected in the pulse-echo method. The second harmonic components in the received waveform in the closed crack area were approximately 6 dB higher than that in the crack-free area. To confirm the origin of the second harmonic components, vibration velocities were detected by using laser Doppler vibrometry. As a result, the threshold amplitude for the generation of the second harmonic components of the Lamb wave was confirmed. This result indicates the existence of the closed cracks in the glass plate.

4:35

4pPA13. Overturning of nonlinear shear waves in media with power-law attenuation. John M. Cormack and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

Simple-wave solutions for nonlinear propagation predict waveforms that develop a vertical tangent at a finite propagation distance and subsequently become multivalued. Attenuation proportional to frequency raised to an exponent greater than unity counteracts nonlinearity sufficiently to prevent the waveform from overturning and becoming multivalued. If the exponent is between zero and unity the waveform may or may not overturn depending on the source amplitude. Multivalued waveforms indicate that the mathematical model lacks an appropriate loss term. The source amplitude above which waveform overturning occurs has been determined as a function of the power-law attenuation exponent for initially sinusoidal waveforms radiated into a medium with quadratic nonlinearity [Cormack and Hamilton, Proc. 22nd Int. Cong. Acoust. (2016)]. The analysis is applied here to nonlinear shear waves propagating in a medium with power-law attenuation using a modified Burgers equation with cubic nonlinearity and a loss term involving a fractional derivative. Introduction of intrinsic coordinates [Hammerton and Crighton, J. Fluid Mech. **252**, 585 (1993)] permits numerical simulation of waveforms that become multivalued. The resulting evolution equation is used to determine the source amplitude above which an initially sinusoidal shear waveform will overturn. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

4:50

4pPA14. Impact non-linear reverberation spectroscopy applied to non-destructive testing of building materials. Nils Ryden (Biomedical Eng., Faculty of Eng., Lund Univ., Box 118, Lund 22100, Sweden, nils.ryden@tg.lth.se), Unn Dahlen (Mathematical Statistics, Faculty of Eng., Lund Univ., Lund, Sweden), Per Lindh (Swedish GeoTech. Inst., Linköping, Sweden), and Andreas Jakobsson (Mathematical Statistics, Faculty of Eng., Lund Univ., Lund, Sweden)

The development of non-destructive testing techniques to detect early micro damage in concrete and other building materials is important for efficient maintenance and management of existing infrastructure. Nonlinear acoustic methods have shown potential for the identification of early damage in brittle materials such as concrete. Commonly, these methods evaluate relative nonlinearity parameters from multiple resonance tests at different amplitudes. We demonstrate a recently developed alternative method, Impact Nonlinear Reverberation Spectroscopy (INRS), where quantitative nonlinearity parameters are evaluated from a single impact resonance test. The recorded reverberation of the measured signal is matched to a synthetic nonlinear damped signal. The proposed model allows instantaneous true physical amplitude, frequency, and damping of each mode to be characterized as a function of time, allowing for quantitative information of the nonlinear parameters. The hysteretic material nonlinearity can be quantitatively characterized over a notably wider dynamic range compared to conventional methods. Two examples from the application to concrete and stabilized soil are presented. In addition quantitative values of the hysteretic nonlinear parameter from stabilized soil samples are compared with the measured ultimate compressive strength of each sample.

5:05

4pPA15. Using elastic nonlinear modulation of sound by sound to detect fracture orientation in reservoir rocks. James A. TenCate (Geophys. Group, Los Alamos National Lab, M.S. D446, EES-17 Earth and Environ. Sci., Los Alamos, NM 87545, tencate@lanl.gov), Alison E. Malcolm (Earth Sci., Memorial Univ., St John's, NF, Canada), and Michael C. Fehler (Earth, Atmospheric and Planetary Sci., Massachusetts Inst. of Technol., Cambridge, MA)

Cracks play a key role in our ability to produce oil and gas or monitor water resources, from microscale cracks that enable permeability in tight formations to larger faults and fractures that compartmentalize reservoirs. Our ability to sense and understand cracks is thus of key importance. We explore the role that cracks play in the nonlinear interaction of propagating waves using an elastic version of modulation of sound by sound. We present a laboratory experiment in which a strong S-wave slightly changes the velocity of a lower amplitude P-wave, and use a rock sample with aligned fractures to demonstrate that this signal is strongly dependent on fracture orientation. We build on the linear slip theory to show that the propagating S-wave is indeed able to open the cracks that the P-wave velocity will be most sensitive to. This gives firm, direct evidence that cracks are a controlling factor in the nonlinear elastic properties of rocks, and opens up the possibility of using such signals to remotely map fracture orientations.

5:20

4pPA16. How does the impact strength affect slow time relaxation in rock. Lev A. Ostrovsky (Appl. Mathematics, Univ. of Colorado, 325 Broadway, R/PSD99, Boulder, CO 80305, lev.a.ostrovsky@noaa.gov) and Andrey Lebedev (Geophys., Inst. of Appl. Phys., Nizhniy Novgorod, Russian Federation)

Earlier we suggested a model for slow time relaxation in rock which is based on the existence of metastable contacts which slowly return to the basic state due to thermal processes. This model allows to explain the logarithmic law of relaxation and estimate the characteristic scale of metastable contacts. In this presentation, along with a brief outline of the problem, we make the next step, namely, describe the dependence of the process on the amplitude of the initial impact. For this, the distribution function of the contacts' strength is used. Based on the known data, we use the lognormal distribution for the thresholds of contact breakups. In the physically evident case of small percentage of breaking contacts, the excited states' concentration is shown to linearly depend the impact stress. The latter is in agreement with the known laboratory experiments. We also used the data from the Parkfield earthquake to evaluate the initial strain level (of the order of 10-5) preceding the relaxation. This estimation seems reasonable as compared with data for earthquakes having the magnitude of M6.

Session 4pPP

Psychological and Physiological Acoustics and Speech Communication: Gap Detection: New Perspectives from Neuroscience, Perception, and Modeling

Shuji Mori, Cochair
Department of Informatics, Kyushu University, 744 Motooka Nishi-ku, Fukuoka 819-0395, Japan

Willy Wong, Cochair
Electrical and Computer Engineering, University of Toronto, 10 King's College Rd., Toronto, ON M5S3G4, Canada

Chair's Introduction—1:00

Invited Papers

1:05

4pPP1. What underlies across-channel gap detection: Overview. Shuji Mori (Dept. of Informatics, Kyushu Univ., 744 Motooka Nishi-ku, Fukuoka, Fukuoka 819-0395, Japan, mori@inf.kyushu-u.ac.jp)

Gap detection, which measures a person's ability to hear gaps in tones and noises, is commonly used to assess auditory temporal resolution. There are a variety of ways to measure the minimum detectable gap duration, or gap threshold, including across-frequency gap detection (detecting a silent gap between two spectrally different markers) and between-ear gap detection (detecting silence between a leading and trailing maker delivered to separate ears). A topic of particular interest concerns across-frequency gap detection, where the gap threshold has been found to be an order of magnitude larger than that obtained for a within-frequency task. The between-ear task has also been demonstrated to yield much longer gap thresholds than the within-frequency task. A number of studies have attempted to resolve the discrepancies between within-frequency and between-frequency/ear tasks (termed jointly as across-channel tasks), and research is still ongoing with a number of different approaches and avenues: psychoacoustic experimentation, electrophysiological measurements at the brainstem and cortical levels, as well as mathematical modeling. In this talk, I will provide an overview of research on across-channel gap detection, including the studies conducted at our laboratory.

1:25

4pPP2. Auditory across-channel processing as assessed by monaural and between-ear temporal gap detection tasks. Ito Kazuhito (Faculty of Information Sci. and Elec. Eng., Kyushu Univ., 744 Motooka Nishi-ku, Fukuoka, Fukuoka 819-0395, Japan, ito@inf.kyushu-u.ac.jp), Takamura Akihide (Graduate School of Information Sci. and Elec. Eng., Kyushu Univ., Fukuoka, Japan), and Shuji Mori (Faculty of Information Sci. and Elec. Eng., Kyushu Univ., Fukuoka, Japan)

Gap detection is a measure of auditory temporal resolution. Hearing people exhibit acute sensitivity to silent gaps between leading and trailing markers when the two markers are identical or similar in frequency. However, sensitivity to such gaps declines when the two markers are of dissimilar frequencies. To examine the hypothesis that auditory gap detection performance declines during temporal comparisons of activity between different perceptual channels, we conducted monaural and between-ear temporal gap detection tasks. In the latter task, the leading and trailing sinusoidal markers delimiting the gap were presented to separate ears, which are regarded as independent channels, at least up to the superior olivary complex in the auditory brainstem. The between-ear gap detection thresholds were increased, even when the two markers were identical in frequency, and elevated gradually as the frequency difference between the two markers became greater than an octave. Furthermore, the patterns of the across-ear gap detection thresholds were roughly comparable with those of the monaural gap detection thresholds, except when the two markers were identical in frequency. These findings suggest that the different "across-channel" processes assessed by monaural and between-ear gap detection tasks share a common central mechanism for managing temporal information originating from different channels.

1:45

4pPP3. Marking time: The precise measurement of auditory gap detection across the lifespan. Jennifer J. Lister (Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, jlister@usf.edu)

Deficits of temporal resolution are thought to contribute to speech understanding in noise difficulties and may be documented using auditory gap detection thresholds (GDTs). It is important to establish the appropriate methods to measure GDTs clinically. We have established GDTs for a variety of stimuli, ages (7-90 years), equipment, degrees of hearing loss, psychophysical paradigms, neurophysiological paradigms, marker relationships (within-channel, across-channel), time points, and presentation ears (left, right, diotic). A number important findings will be discussed: 1. Best stimulus for measurement of GDTs is narrow-band noise. 2. GDTs improve from ages 7 to 9, stabilize between ages 9 and 40, and deteriorate with age thereafter. 3. GDTs may be measured reliably using a variety of

equipment. 4. Hearing loss has a minor impact on GDTs. 5. A 2-interval psychophysical paradigm is best for measuring GDTs. 6. GDTs may be documented using the P1-N1-P2 auditory evoked potential. 7. Across-channel GDTs provide different information than within-channel GDTs. 8. GDTs are reliable within and across test sessions. 9. GDTs do not differ across ear conditions. A stable and sensitive measure of temporal resolution that may be used in a clinical setting to assess temporal resolution will be recommended.

2:05

4pPP4. Auditory evoked fields elicited by repetitive gaps embedded within a continuous tone. Hidehiko Okamoto (Dept. of Integrative Physiol., National Inst. for Physiological Sci., 38 Nishigo-Naka, Myodaiji, Okazaki 4448585, Japan, hokamoto@nips.ac.jp)

The N1m response, which is a prominent auditory evoked component with a latency of around 100 ms, elicited by sound onset is known to decrease with the repetition of identical sound stimuli. Previous studies demonstrated that a silent gap embedded within a continuous sound could also elicit the N1m responses; however, it remains elusive whether the N1m decrement occurs with a repetition of silent gaps. In the present study, we investigated the decrement of N1m responses elicited by five repetitive pure tones presented in silence and by five repetitive silent gaps embedded within a continuous tone using magnetoencephalography. The results obtained demonstrated that the N1m responses elicited by the repetitive pure tones decreased from the 1st to the 2nd and remained constant from the 2nd to the 5th, whereas those elicited by the repetitive gaps in a continuous tone gradually decreased from the 1st to the 5th. Those results indicate that neural refractoriness appears to mainly contribute to the decrement of the N1m responses elicited by repetitive pure tones, while habituation may play an additional role in the decrement of the N1m responses elicited by repetitive gaps in a continuous tone.

2:25

4pPP5. Temporal asynchrony and rhythm perception: Neural bases of perceptual asymmetries. Andrew J. Oxenham and Magdalena Wojtczak (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Temporal gap detection has been proposed as a perceptual and neural measure of temporal resolution. This measure has not shown strong frequency dependence but, if anything, high frequencies have been found to produce slightly lower (better) thresholds than very low frequencies. In apparent contradiction to these findings, a recent study claimed superior time perception for lower frequencies and proposed the finding as an explanation for why low-frequency instruments typically carry the rhythm in music [Hove *et al.* (2014), Proc. Natl. Acad. Sci. U.S.A. **111**, 10383-10388]. The present study reviewed those findings and replicated them, but found that the results in fact reflect an asymmetry in asynchrony detection, reported earlier, whereby asynchronies are more readily detected when the lower-frequency tone is leading than when the higher-frequency tone is leading [Wojtczak *et al*., (2012). J. Acoust. Soc. Am. **131**, 363-377]. New results using behavioral and EEG approaches reconcile the data from these different studies and confirm that low frequencies do not provide superior temporal resolution. Instead they suggest an asymmetry in how onset asynchronies are perceived, which may have its roots in the acoustic properties of natural sounds and in the mechanics of cochlear filtering. [Work supported by NIH grant R01DC005216.]

2:45

4pPP6. A peripheral model of gap detection. Willy Wong and Hugo Lepage (Elec. and Comput. Eng., Univ. of Toronto, 10 King's College Rd., Toronto, ON M5S3G4, Canada, willy.wong@utoronto.ca)

Gap detection is an important measure of the temporal resolution of the auditory system. We present a model of gap detection involving the statistics of peripheral auditory neural activity: detecting a gap involves the detection of an interval that is different from the usual interspike interval found in the auditory spike train. The model is capable of resolving a number of important observations in empirical gap research including why across-channel (or across-frequency) gap detection is an order of magnitude larger than gap detection performed in channel (i.e., using the same frequency). We achieve this by recognizing that in-channel detection involves a different strategy than across-channel detection. The model also provides an explanation for why the intensity dependence of gap detection is limited to a 40 dB range, and why critical bands are crucial towards defining the notion of a channel in gap detection. Specific predictions can be made about the shapes of the psychometric functions.

3:05–3:25 Panel Discussion

Session 4pSA

Structural Acoustics and Vibration, Physical Acoustics, and Noise: Fluid Flow Induced Vibration and Noise

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

Kuangcheng Wu, Cochair
Ship Survivability, Newport News Shipbuild, 4101 Washington Ave., Newport News, VA 23693

Chair's Introduction—1:00

Invited Papers

1:05

4pSA1. A trial application of acoustic excitation for jet noise reduction. Shigehiko Kaneko (Dept. of Mech. Eng., The Univ. of Tokyo, 7-3-1, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, kaneko@mech.t.u-tokyo.ac.jp) and Tatsuya Ishii (Aeronoutical Technol. Directorate, Japan Aerosp. Exploration Agency, Tokyo, Japan)

Jet noise, one of the dominant noise components from an aircraft jet engine is known as the result of developed vortices produced during jet mixing process. In last couple of years, a method for jet noise reduction which acoustically controls the jet mixing layer using a sound excitation is proposed and investigated under the collaboration between The University of Tokyo and JAXA. In this paper, the main idea, experimental apparatus, experimental condition and experimental results will be explained. The experiment was conducted under an unheated and subsonic jet condition. The results show that low frequency sound excitation results in an increase in low frequency noise and a decrease in high frequency noise. The low frequency sound excitation is considered to excite the instability of jet flow and collapse the jet plume. On the other hand, the high frequency sound excitation resulted in a slight increase of high frequency noise and a slight decrease of low frequency noise (0.5 dB reduction in maximum) similar to the effect of Chevron nozzle. In addition, LES simulation was conducted to investigate the effect of mixing process caused by sound excitation.

1:25

4pSA2. Simultaneously modeling of multiple fluid-structure interaction subsystems of the human vocal folds. Ryan M. Oakey, Eduardo J. Alvarez (Mech. Eng., Brigham Young University-Idaho, Rexburg, ID), and Scott L. Thomson (Mech. Eng., Brigham Young Univ., 435 CTB, Provo, UT 84602, thomson@byu.edu)

Voice production is a direct result of vocal fold flow-induced vibration. Air from the lungs dynamically interacts with vocal fold tissue to generate self-sustained vocal fold oscillation, which in turn creates a fluctuating pressure field that is the source of sound for voice and speech. In addition to this primary air-tissue interaction that drives vocal fold vibration, other fluid-structure interactions (FSIs) occur within the interior regions of the vocal folds. For example, during voicing, the vocal fold vasculature is subjected to a time-varying displacement field as the tissue vibrates, thereby potentially affecting blood flow dynamics. Additionally, in benign vocal fold lesions comprised of fluid-filled pockets within the vocal folds, known as polyps, there are significant interactions between the fluid within the polyps and the surrounding tissues. The vocal fold system, therefore, potentially consists of multiple subsystems with different types of fluid-structure interactions that are distinct, yet coupled. In this presentation, experimental and computational studies that have focused on simultaneous modeling of these multi-FSI subsystems will be discussed. Methodologies for modeling the various FSI components will be described, sample results will be presented, and current capabilities and limitations of the various approaches will be summarized.

1:45

4pSA3. Acoustic emission due to partial cavity shedding on a hydrofoil. Juliana Wu (Naval Architecture and Marine Eng., Univ. of Michigan, 2010 Beal Ave., Ann Arbor, MI 48109, wuju@umich.edu), Harish Ganesh (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), and Steven Ceccio (Naval Architecture and Marine Eng., Univ. of Michigan, Ann Arbor, MI)

The partial cavity shedding dynamics on a NACA 0015 hydrofoil were examined at angles of attack of 7 and 10 degrees over a range of cavitation numbers. Hydrophone measurements in conjunction with high-speed videos and time resolved X-ray densitometry were used to acquire time synchronous measurements of the void fraction field of the cavity and the emitted noise. Cavitation dynamics changed significantly with the reduction of the inlet cavitation number. At higher cavitation numbers, small stable partial cavities were observed. Upon reduction of cavitation number, the cavities grew in length and pinched off from the rear of the cavity, due to a liquid re-entrant jet. Upon further reduction of inlet cavitation number, the dynamics changed significantly with the cloud collapse of the shed

vapor inhibiting the cavity growth. X-ray densitometry measurements revealed the presence of a propagation bubbly shockwave as a mechanism of shedding at the lowest cavitation numbers. This process caused complex, multi-step cavity dynamics. Spectral analysis of the acoustic signal showed there were multiple acoustic frequency peaks, instead of a single well-defined shedding frequency. Spectral content of the acoustic measurements was then analyzed to identify different flow processes responsible for their generation.

2:05

4pSA4. Experimental investigation of compliant wall deformation in a turbulent channel flow. Cao Zhang, Jin Wang, and Joseph Katz (Dept. of Mech. Eng., Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218, katz@jhu.edu)

Interaction of a compliant wall with a turbulent channel flow is investigated by simultaneously measuring the time-resolved, three-dimensional flow field using tomographic PIV and the two-dimensional surface deformation using Mach-Zehnder interferometry. The friction Reynolds number is $Re_\tau = 2300$, and the Young's Modulus of the wall is 0.93 MPa, resulting in a ratio of shear speed to center-line velocity (U_0) of 6.8. The wavenumber-frequency spectra of deformation contain a non-advected low-frequency component and advected modes, some traveling at U_0 and others at $0.72U_0$. The wall dynamics is elucidated by correlating the deformation with flow variables, including the 3D pressure distribution. The pressure-deformation correlations peak at y/h~0.12 (h is half channel height), the elevation of Reynolds stress maximum in the log-layer. Streamwise lagging of the deformation behind the pressure is caused in part by phase-lag of the pressure with decreasing elevation, and in part by material damping predicted by the Chase (1991) model. Positive deformations (bumps) are preferentially associated with ejections involving spanwise vortices downstream and quasi-streamwise vortices with spanwise offset, consistent with presence of hairpin-like structures. The negative deformations (dents) are preferentially associated with a positive pressure fluctuation at the sweep-ejection transition. [Sponsored by ONR.]

Contributed Papers

2:25

4pSA5. Structural vibration of an elastically supported plate due to excitation of a turbulent boundary layer. Jonmarcos Diaz, Kevin Maki, and Nickolas Vlahopoulos (Naval Architecture and Marine Eng., Univ. of Michigan, 2600 Draper Dr., Ann Arbor, MI 48109-2145, jonmarco@umich.edu)

High-Reynolds number turbulent boundary layers are an important source for noise and structural vibration. Indeed small features of a structure can have important influence on the resulting noise and vibration. In this work we develop a new method to couple a high-fidelity fluid solver with a dynamic-finite-element solver for the structure. The flow solver is based on the OpenFOAM opensource CFD toolkit. The fluid solver uses the Large-Eddy Simulation closure for the unresolved turbulence. Specifically, a local and dynamic one-equation eddy viscosity model is employed. The fluid pressure fluctuation on the structure is mapped to the dynamic finite element model with a suitable interpolation routine. A modal decomposition of the dynamic finite-element model is used to reduce the number of degrees-of-freedom of the numerical structural model. The numerical method is validated for the turbulent flow over a flat plate. The plate is elastically supported along its edges, and the turbulence is excited upstream of the plate by the means of a solid obstruction. Results for the structural vibration on the plate will be compared with previously published experimental measurements. Furthermore, the results will be compared with other numerical predictions in which the fluid forcing is determined using statistical methods

2:40

4pSA6. Effects of vocal tract inertance on the glottal flow. Charles P. Farbos de Luzan, Liran Oren (Dept. of Otolaryngol. - HNS, Univ. of Cincinnati, Medical Sci. Bldg., Rm. 6303B, Cincinnati, OH 45267-0528, farboscs@mail.uc.edu), Ephraim Gutmark (Dept. of Aerosp. Eng., CEAS - Univ. of Cincinnati, Cincinnati, OH), and Sid Khosla (Dept. of Otolaryngol. - HNS, Univ. of Cincinnati, Cincinnati, OH)

Compressible large eddy simulation is employed to numerically investigate the laryngeal flow. One static model of the canine larynx with a divergent glottis is considered, with the presence of false vocal folds (FVFs). This computational model is developed from empirical data, and compared to similar configurations that do not involve the subject-specific geometrical features. Due to the high enough Reynolds number, the flow is unsteady and develops asymmetric states downstream of the glottis. The intra-glottal vortex structures are formed on the divergent wall of the glottis, immediately downstream of the separation point. The vortices are then convected downstream and characterized by a significant negative static pressure. The FVFs are a main factor in the generation of stronger vortices, and thus on the closure of the TVFs. Models with and without divergent vocal folds are investigated and linked to the existence of vortices. The direct link between the FVFs geometry and the motion of the TVFs, and by extension to the voice production, is of interest for medical applications as well as future research works.

2:55

4pSA7. Noise source identification and noise reduction in a multi-state, centrifugal air compressor system. Je-Heon Han (Mech. Eng., Korea Polytechnic Univ., 237. Sangidaehak-ro, Siheung-si, Gyeonggi-do 15073, South Korea, jeep2000@kpu.ac.kr), Yaying Niu (Dept. of Mech. Eng., Texas A&M Univ., College Station, TX), Changwoo Lim (R&D Ctr., Hanwha Techwin, Seongnam-si, South Korea), and Yong-Joe Kim (Dept. of Mech. Eng., Texas A&M Univ., College Station, TX)

The objective of this research is to recommend practical noise control schemes for Hanwha Techwin's multi-stage, centrifugal air compressors. For purpose of achieving this objective, far- and near-field noise measurements, sound intensity measurements, and structural vibration measurements were performed first to identify the acoustic characteristics of a compressor. Through these measurements, it was shown that the most dominant noise and vibration components were at the blade passing frequencies (BPFs) of the compression stage impellers. Thus, by suppressing these tonal BPF noise components, the overall noise level was reduced effectively. Here, Helmholtz resonators, dynamic dampers, noise insulation materials, and an enclosure were recommended to reduce these tonal noise components. The Helmholtz resonators could be used to reduce the overall SPL up to 3.5 dBA. By applying the mass dampers, 2.7 dBA reduction at 7 kHz and 1.5 dBA reduction at 10 kHz could be achieved. The installation of the sound insulation material could achieve 8 dBA overall SPL reduction. By installing the enclosure around the compressor, 4.3 dB reduction in the overall SPL could be achieved. To achieve a higher noise reduction using the enclosure, the size of leakage holes between the compressor pipe and the enclosure is recommended to be reduced.

Session 4pSC

Speech Communication: Non-Native Speech Production and Perception, Learning, and Adaptation (Poster Session)

Nicole Holliday, Chair
Linguistics and Cognitive Science, Pomona College, Pomona College, Edmunds 185, 185 E Sixth St., Claremont, CA 91711

All posters will be on display from 1:00 p.m. to 5:30 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:15 p.m. and authors of even-numbered papers will be at their posters from 3:15 p.m. to 5:30 p.m.

Contributed Papers

4pSC1. The influence of long-term exposure on the perception of foreign-accented speech. Hayk Abrahamyan (Psych., SUNY at Buffalo, Univ. at Buffalo, Park Hall 204, Buffalo, NY 14260, hayk@buffalo.edu)

We recognize spoken words in our native language with relative ease. Yet, difficulties arise when we encounter a talker with an accent. Short-term lab exposure leads to observable improvements in comprehension of foreign-accented speech. Life-long exposure markedly improves perception of within language variability (i.e. dialects) and refines listeners' representations. Less is known about the benefits of long-term exposure to foreign-accented speech. We used cross-modal identity priming to investigate if long-term exposure to Chinese-accented English by native speakers of American English would modulate their processing of Chinese-accented speech. We predicted that listeners with long-term exposure to the foreign accent, assumed based on long-term residence in the New York City area, should be faster and more accurate at recognizing Chinese-accented words than listeners without long-term exposure, assumed based on long-term residence in less linguistically diverse areas. Our preliminary results present a surprising finding indicating that the assumed long-term exposure had the opposite effect, where the listeners with long-term exposure to a foreign accent are unexpectedly slower and less accurate than the listeners without long-term exposure. Our results suggest that long-term exposure does affect listeners' abilities to recognize and represent foreign-accented spoken words.

4pSC2. Segmental versus prosodic predictors of perceived accentedness. Elizabeth A. McCullough and Richard Wright (Linguist, Univ. of Washington, Box 352425, Seattle, WA 98195, emccullo@uw.edu)

Previous work has identified both segmental and prosodic influences on the perceived foreign accentedness of non-native speakers, but their relative contributions remain poorly understood, in part because they are often investigated independently. In this study, both segmental and prosodic properties of word-length stimuli were assessed. Native English-speaking listeners rated the accentedness of trochaic English words produced by talkers from four language backgrounds: Hindi, Korean, Mandarin, and Spanish. Each token's deviation from native English productions was quantified for two segmental properties (VOT and vowel quality) and three prosodic properties (ratios of duration, intensity, and f0 across the two syllables). For each language background, a linear mixed-effects regression model was created to predict accentedness ratings from the phonetic deviations, and the significance of each fixed effect was examined. In each model, the significant predictors included both segmental and prosodic properties. For L1 Hindi and L1 Spanish talkers, the most influential predictor was segmental; however, for L1 Korean and L1 Mandarin talkers, the most influential predictor was prosodic. Thus, even for short stimuli, both segmental and prosodic information must be considered in accounting for accentedness judgments. Listeners are sensitive to different ways that foreign accent may be manifested across different non-native backgrounds.

4pSC3. Recognition of foreign-accented words in isolation. Mirrah Maziyah Mohamed (Univ. at Buffalo, Singapore, Singapore, Singapore) and Hayk Abrahamyan (Univ. at Buffalo, 245 Park Hall, Buffalo, NY 14260, hayk@buffalo.edu)

Foreign-accented speech presents a challenge for native speakers of a given language. When presented with foreign-accented speech, native speakers tend to rely heavily on context and other lexical cues. When processing isolated spoken words, participants do not have access to contextual cues and have to rely solely on acoustic cues. Thus, processing accented spoken words in isolation might be a bigger challenge. We tested the hypothesis that identification accuracies by native speakers of American English of two foreign-accented speakers with different accent strengths will not differ in isolated spoken word recognition task. Ten native speakers of American English rated the accent of a Chinese-accented (heavy accent) and a Korean-accented (moderate accent) speakers after hearing a small passage read by both speakers. Different set of native speakers of American English heard a list of isolated words spoken by both speakers and had to type what they had heard. Overall, participants did not differ in their accuracy in the identification task even though the speakers differed in the strength of their accent, while there was no significant difference in participants rating of the strength of the speakers' accents. Implication of the results on the processing of foreign-accented speech will be discussed.

4pSC4. Speaker similarity, acoustic properties, and perceptual learning with non-native speech. Hanyong Park (Dept. of Linguist, Univ. of Wisconsin-Milwaukee, Johnston Hall 123, P.O. Box 413, 2522 East Hartford Ave., Milwaukee, WI 53211, park27@uwm.edu) and Noah H. Silbert (Dept. of Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH)

Variability in speech signals plays an important role in perceptual learning of non-native speech. Between-talker differences are a major source of variability. The present work aims to quantify within- and between-talker variability, following the approach of Silbert & Park (2014, *JASA*, **136**, 2714) to quantify variation within and across Korean stop categories. We analyze 20 (10 male and 10 female) native speakers' repeated productions of the Korean CVC nonword /p'ap/, using statistical techniques for dimensionality reduction to map multidimensional acoustic space to lower-dimensional talker-similarity space. We then compare the modeled similarity space to speaker discrimination data. Data were collected in an oddball task, where the listeners choose the oddball after listening to a set of triplets consisting of /p'ap/s spoken by two speakers (two distinct tokens from one speaker, one token from the other). The modeled similarity space allows us to automatically select tokens based on talker similarity and provides a rigorous quantitative method for assessing relationship between multidimensional acoustic properties and perceived speaker similarity. This approach promises to provide a rigorous, automated method for quantitatively defining and manipulating variability in perceptual learning experiments.

4pSC5. The role of vowel quality in the perception of English lexical stress for native speakers of Japanese. Sayoko Eguchi (Kobe Univ., ATR, Nada-ku tsurukabuto 1-2-1, Kobe-shi, Hyougo-ken 657-8501, Japan, syk.eguchi@gmail.com)

Vowel quality has been hypothesized to be a cue in determining the location of stress in English words. This study examined the role vowel quality plays in perception by native speakers of Japanese and English using synthesized English words: *conflict*, *contract*, and *content*. Suprasegmental (SP) parameters—F0, duration, and intensity—and vowel quality (VQ) parameters—the frequencies of the first two formants, were independently manipulated to make varying versions of the stimuli. Results of the perception experiment showed that native speakers of Japanese identified the stress location more accurately in the SP-driven stimuli, in which the stress pattern was generated using SP parameters, than in the VQ-driven stimuli, in which the stress pattern was generated using VQ parameters. In contrast, native speakers of English did not differentiate between SP-driven stimuli and VQ-driven stimuli. Furthermore, for the stimuli that SP parameters and VQ parameters indicated incongruent syllable, Japanese speakers identified the stress as located on the SP-driven syllable more often than English speakers did. These results demonstrated that native speakers of Japanese do not use VQ as a cue to perceive stress location whereas native speakers of English do use VQ as a cue. [Work supported by JSPS KAKENHI 23242032.]

4pSC6. Enlglish-as-a-second-language speakers' perception of personality in English speech. Bin Li, Yingting Cui, Yan Dou (Dept. of Linguist and Translation, City Univ. of Hong Kong, Kowloon Tong 000, Hong Kong, binli2@cityu.edu.hk), and Yan Liu (Computing, the Hong Kong Polytechnic Univ., Hong Kong, Hong Kong)

We make inferences on speaker's personality and emotion from their voices. This study aims at English-as-a-second-language (ESL) speakers' perception of personality as reflected in public speech online given by native English speakers. Phonetic parameters such as speaking rate and pitch are manipulated and combined in different ways to examine their effects on perceptual judgement of personality and content comprehension. Our ESL listeners are 50 university students who are native speakers of Cantonese and fluent users of English. They answer questions on the content of the speech samples, and also make judgement on various traits of the speaker' personality. The correct percentage and the judgement scores are recorded and examined for correlation between manipulation types and listening comprehension as well as that between phonetic parameters and perception of personality. Modification to the speaker's pitch seem in general improve listeners' comprehension accuracy as well as results in higher scores of personality perception, whereas that to the speaking rate does not show positive influence on speech comprehension or personality judgment. Our findings will provide insights to general phonetics and speech communication, but also pedagogical implications to academic English in nonnative settings.

4pSC7. Perception of speech rate in speech rate perception. Yahya Aldholmi and Hanyong Park (Dept. of Linguist, Univ. of Wisconsin-Milwaukee, Johnston Hall, Rm. 117, 2522 East Hartford Ave., Milwaukee, WI 53211, aldholmi@uwm.edu)

This study investigates what influences a listener's perception of speech rate. Three factors were investigated: pause type, actual speech rate, and utterance type (native, accented, and unfamiliar). Using manually accelerated vs. decelerated but structurally-identical sentences with three-type pauses (long, short, and filled) in accented, native, and unfamiliar language (Spanish) utterances, a group of Arabic speakers judged whether the sentences they heard were slow or fast, and rated the speed on a 1-7 scale (1 = slow, 7 = fast). The findings show that long pauses triggered lower ratings, while filled pauses achieved ratings higher than both long and short pauses. Unfamiliar language received the highest ratings while the accented speech received the lowest. This suggests that, regardless of actual speech rate, silent pauses affect our perception of speech rate, and that when the duration of silence differs, our perception of speech rate will differ accordingly. The results also show that we tend to think of accented speech as being slower and of unfamiliar speech as being faster even when the two speech samples have the same length and pause types.

4pSC8. English listeners' judgments of Japanese-accented English vowel productions: Relation to acoustic and kinematic characteristics. Sonya Mehta and William F. Katz (The Univ. of Texas at Dallas, Callier Ctr., 1966 Inwood Rd., Dallas, TX 75235, naya@utdallas.edu)

The present study investigates how monolingual English listeners categorize foreign (Japanese) productions of English tense and lax vowels with the aim of relating vowel identification to the acoustic and kinematic vowel space. American English front vowels (/i/, /ɪ/, /e/, /ɛ/, and /æ/) were elicited in /hVd/ context from ten Japanese adults who learned English as a second language (ESL). Listeners categorized these vowel productions in a six-alternative, forced-choice paradigm (including the choice /ʌ/). Vowels produced by two native English speakers were included in the perceptual task for comparison. Preliminary results showed higher perceptual accuracy for tense vowels than for lax vowels. As expected, vowels from English talkers were more accurately identified than those from Japanese ESL speakers. To determine whether listeners' perceptual categorization corresponded with acoustic and kinematic patterns, perceptual data for contrasting vowel categories were compared with measured differences in formant (F1/F2) and articulatory space. Pilot data suggest a strong relation between acoustic and kinematic measures, but a weaker correspondence between these two data sources and listeners' perceptual judgments. Additional factors, including the roles of duration, talker and listener variability, and vowel similarity between languages, will be explored.

4pSC9. Non-native speakers perform poorer than native speakers in Japanese speech recognition in reverberation. Eri Osawa, Takayuki Arai (Sophia Univ., 7-1, Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, eri1989.16.11@gmail.com), Nao Hodoshima (Tokai Univ., Tokyo, Japan), and Takako Igeta (Sophia Univ., Tokyo, Japan)

Reverberation affects perceptual performance, especially non-native listeners, elderly people, and people with hearing impairments. Native Japanese speakers confused some Japanese consonants and they failed to distinguish Japanese length contrast in reverberation [Arai *et al.*, Proc. Autumn Meet. Acoust. Soc. Jpn., 2016]. The current study examined degradation degree of non-native speakers (native English speakers) in Japanese speech recognition in reverberation, using the same experiment settings with the previous study. There were two sets of stimulus: 1) Japanese consonant-vowel (CV) syllables, 2) non-words varying in duration of a certain vowel/consonant along durational continuum. In the results, although non-native speakers confused consonants even in non-reverberant condition, the number of their confusion in reverberation was larger than that in non-reverberant condition. Confusion of non-native speakers in reverberation was much more than that of native speakers. In addition, the results indicated that non-native speakers could hardly distinguish the length contrast of Japanese in reverberation. The proportion of "long" responses did not change sharply in the perception of vowel continuum, while native speakers changed their responses at some points on the continuum. The results suggest that learners need training for listening to speech sounds in reverberation.

4pSC10. Transferring cue-weighting from second language into first language: Group trends and individual differences. Olga Dmitrieva, Jenna Conklin (Purdue Univ., 640 Oval Dr., Stanley Coulter 166, West Lafayette, IN 47907, odmitrie@purdue.edu), and Yulia Nigmatulina (Saint-Petersburg State Univ., Saint-Petersburg, Russian Federation)

Second language learning has been shown to affect first language production (Flege, 1987; Major, 1992; Sancier & Fowler, 1997, inter alia). Less is known about whether and how experience with another language may affect first language perception. The present study examines the use of preceding vowel duration vs. voicing during closure as cues to word-final and word-medial stop voicing by three groups of listeners: native speakers of American English (34), native speakers of Russian (34), and Russian expatriates in the United States (24). The results suggest that bilingual listeners are transferring a greater reliance on vowel duration, characteristic of English perceptual mode, into native speech categorization. The transfer occurred in word-final condition where vowel duration as a correlate of voicing is of very different importance in the two languages. Individual trends revealed that bilinguals' reliance on vowel duration may even be

exaggerated in native, compared to non-native, speech perception. The use of vowel duration in bilinguals was positively correlated with such individual differences as length of residence in the United States and average amount of speaking English per week. The triggers and possible benefits of such perceptual transfers are discussed in the context of language use in the expatriate communities.

4pSC11. Categorical perception of Mandarin tones by Chinese-native, English-native and Chinese-as-a-second-language English listeners. Wenyi Ling, Theres Grüter (Second Lang. Studies, Univ. of Hawaii at Manoa, 1711 East West Rd., 614B, Honolulu, HI 96848, wenyi9@hawaii.edu), and Amy Schafer (Linguist, Univ. of Hawaii at Manoa, Honolulu, HI)

Previous research found that native-Chinese listeners perceive tones more categorically than listeners with no knowledge of a tonal language (naïve listeners; Hallé *et al*., 2004). This study examined tone perception by 26 native-English adult learners of Mandarin (L2ers) in comparison to 30 naïve and 30 native-Chinese listeners. Identification and AXB discrimination tasks tested monosyllables (/pi/, /pa/) with 9-step F0 continua between all possible Mandarin tone pairs. Native listeners showed more categorical identification (steeper slopes) than naïve listeners. The L2 group showed significantly shallower identification slopes than native listeners ($p<0.01$) and steeper slopes than naïve listeners ($p<0.01$). L2 proficiency (listening test plus self-report) positively correlated with identification performance ($r=0.36$, $z=-2.56$, $p<0.01$), suggesting higher proficiency may lead to more native-like tone identification. However, although L2ers' discrimination accuracy (0.90) was significantly higher than native listeners' (0.82; $z = -4.76$, $p<.001$), it did not differ significantly from naïve listeners' (0.90; $z = -0.61$, $p=0.54$), suggesting that the non-native groups discriminated tone pair continua in a similar way. Tone pairs involving level tones (e.g., T1-T3) showed more categorical patterns than those involving two contour tones (e.g., T2-T3) in each group, consistent with the Anchor Hypothesis (Xu *et al*., 2006).

4pSC12. Reaction time of Japanese listeners to retroflex and bunched /r/ pronunciation by native English speakers. Mai Gunji (Univ. of Aizu, Tsuruga, Ikki-mati, Aizuwakamatsu, Fukushima 965-8580, Japan, s1210154@u-aizu.ac.jp), Ian Wilson, and Jeremy Perkins (Univ. of Aizu, Aiduwakamatu, Japan)

In this study, we focus on Japanese learners' reaction time (RT) to retroflex and bunched pronunciation of /r/ in English words spoken by native English speakers. In junior high school, Japanese students generally learn only retroflex pronunciation of /r/. If there is a strong link between production and perception, we would expect those students to be able to perceive retroflex /r/ faster than bunched /r/. We carried out a forced-choice RT experiment for 30 native Japanese listeners and 4 native English controls. This experiment used 2 speakers' voices (both Canadian English) and 9 minimal pairs of /r/ and /l/ words. Stimuli were spoken words and picture-pairs (two simultaneously presented in each trial). Listeners had to identify the spoken word by choosing the left or right picture. As a result, we could measure whether it is easier to perceive sounds pronounced the same way you speak or not. From the results, we found that the RTs for retroflex and bunched pronunciation of English words spoken by native speakers were not significantly different, even for native listeners. In addition, overall accuracy rates were very low among the Japanese speaking participants (66.3%, compared to 99.7% for the English speaking participants).

4pSC13. Phonotactic knowledge and phonetic details in the perception of non-native consonant sequences: Evidence from Mandarin Chinese. Qianwen Guan and Harim Kwon (Université Paris Diderot, Bâtiment Olympe de Gouges 8 Pl. Paul-Ricoeur, Paris 75013, France, qianwen_guan@hotmail.com)

Previous research has shown that non-native consonant sequence perception is influenced by both the phonotactic knowledge of native language and the phonetic details of the stimuli. To further examine the effects of these two factors, we conducted two perception experiments with monolingual Mandarin listeners. In experiment 1, the listeners determined whether they heard a vowel between two consonants. Stimuli were VC_1C_2V with various intervocalic consonant sequences produced by a Russian speaker. The listeners reported hearing a vowel between two consonants (56%), providing evidence for the effects of native phonotactics. We also found that duration of C1 release influenced the perception of epenthetic vowel. In experiment 2, the same participants heard the non-native sequences in word-initial and intervocalic positions and transcribed them in Pinyin. 75% of the transcription data showed a vowel epenthesis. In addition, different strategies were observed in different positions in a word: word-initial position showed more C1 deletion presumably due to weak acoustic cues of C1 and intervocalic position showed more C2 deletion. Taken together, these findings indicate that while the native phonotactics play a dominant role in Mandarin listeners' perception of non-native consonant sequences, the effects of phonetic details of the stimuli are not negligible.

4pSC14. Lexical age of acquisition effects on native and non-native speech perception. Renee Kemp (Linguist, Univ. of California, Davis, 469 Kerr Hall, UC Davis, Davis, CA 95616, rlkemp@ucdavis.edu)

Lexical difficulty has been shown to play a role in both speech production and perception. The relative difficulty of a word can be determined based on lexicon properties such as usage frequency, neighborhood density, or lexical Age of Acquisition (AoA). These factors have been shown to predict a range of different acoustic modifications such as hyperarticulation and increased duration. The current study investigates the effect of AoA-conditioned phonetic variation in both plain and foreigner-directed speech conditions on both native and non-native English speech perception. 60 total subjects completed a lexical decision task. Investigating these two populations allows for an examination of the role of lexical difficulty in language acquisition. The data show that lexical AoA influences the accuracy of lexical decision responses. Specifically, native speakers were more accurate in their identification of words with a low AoA rating compared to a high AoA rating (i.e. more lexically difficult items). Interactions between AoA and speech condition were also observed. The findings of this study suggest acoustic modifications can serve, at least in part, to facilitate the perception of lexically difficult words.

4pSC15. Comparative acoustic analyses of non-native English: The search for systematic variation. Rebecca Laturnus (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, rcl309@nyu.edu)

Previous research has shown that exposure to multiple foreign accents facilitates adaptation to an untrained novel accent (Baese-Berk *et al.* 2013). The explanation offered is that L2 speech varies systematically, such that there are commonalities in the productions of non-native speakers, regardless of their language background (e.g. L2 speech is slower and many of English's contrasts are difficult for speakers of other languages). The current work conducted a systematic acoustic comparison between two native English speakers and six non-native accents that closely matched those used in Baese-Berk *et al.* (2013). All talkers, taken from the Wildcat Corpus, were male, and the non-native voices had comparable foreign accentedness ratings (Van Engen *et al.* 2010). VOT and formant values of stressed and unstressed vowels were analyzed, comparing each non-native accent to the native English talkers. Additionally, pairwise variability indices were calculated for vocalic and consonantal intervals as measures of rhythm. The results for each measure show substantial variability across speakers, reflecting phonetic transfer from individual L1s, rather than commonalities in their productions. The data are therefore more consistent with a hypothesis of accent attunement wherein variability causes listeners to relax their expectations of what constitutes a good category exemplar.

4pSC16. The effects of prenasal raising of American English /æ/ on the identification of American English vowels by native Japanese and Korean listeners. Takeshi Nozawa (Lang. Education Ctr., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, t-nozawa@ec.ritsumei.ac.jp) and Sang Yee Cheon (Dept. of East Asian Lang. & Literatures, Univ. of Hawaii at Mānoa, Honolulu, HI)

This study attempts to investigate the effect of prenasal raising of American English /æ/ on the identification of the vowel by native Japanese and

Korean listeners by comparing the identification accuracy of /æ/ in preplosive and prenasal context by these two listener groups. Native Japanese speakers generally equate /æ/ with Japanese low vowel /a/ whereas native Korean speakers identify /æ/ with Korean mid front vowel /e/, and /æ/-/ɛ/ is one of the most difficult English vowel pairs for Korean speakers to discriminate (Ingram & Park 1997, Frieda & Nozawa 2007). It was expected that Korean listeners would identify raised /æ/ in prenasal context as /ɛ/ or /eɪ/, but they mistakenly identified /ɛ/ and /eɪ/ as /æ/ far more frequently than /æ/ as /ɛ/ or /eɪ/. The results revealed that while Japanese listeners' identification accuracy of /æ/ is significantly lower in prenasal context than in preplosive context, Korean listeners' identification accuracy of /æ/ is unaffected by the raising of /æ/. Korean listeners far more often mistakenly identify /ɛ/ as /æ/ in preplosive context than the other way around, and this tendency stays the same even when /æ/ is raised in prenasal context.

4pSC17. The effects of second language experience on the perception of native language prosody. Grace E. Oh (English Lit. and Lang., Konkuk Univ., 120 Neungdong-ro, Gwangjin-gu, Seoul 05029, South Korea, gracey1980@yahoo.com)

The effects of second language experience on adult learners' perception of native language prosody were investigated. A total of 39 sentences produced in Korean, Mandarin, Japanese by three female speakers were low-pass filtered and presented to three groups of 10 participants with different language backgrounds: Native Korean speakers with no L2 experience (NK group), native Mandarin speakers (NM group) and Korean learners of Chinese (KC group) with an average LOR in China of 1 year. The participants were instructed to listen to each stimulus and decide whether it was Korean or not based on the suprasegmental features. Overall, the NK group (76%) revealed significantly higher discrimination accuracy than the KC (65%) or the NM group (57%). However, the NM participants with extensive Korean experience showed significant improvement in accuracy for both Korean (75%) and Chinese (84%). The results, on the one hand, support the significant effect of L2 experience on the acquisition of L2 prosody. The KC group's lower accuracy for Korean, on the other hand, suggests that too much attention may have been directed towards acoustically salient L2 cues (e.g., F0 range) in the perception of L1 prosody.

4pSC18. English spoken word recognition for Japanese native speakers: A comparison with online and offline processing. Yoko Sakamoto (Premedical Sci., Dokkyo Medical Univ., 880 Kitakobayashi, Mibu, Shimotsugagun, Tochigi 3210293, Japan, y-saka@dokkyomed.ac.jp)

The aim of this study was to investigate the mechanisms of English (L2) spoken word recognition for Japanese native speakers by two experiments using a cloze test. In Experiment 1, 90 Japanese university students were divided into three groups based on their listening level. Then they took a listening test of a CNN material. The correct rates and error patterns were analyzed. The result showed that the correct rates for three groups were less than 50%, however, the upper group took the highest correct rate and the lowest number of blanks, extra phoneme insertions and phoneme recognition errors. In Experiment 2, 32 Japanese university students were divided into two groups, and one group took a cloze test without listening, and the other took the cloze test with listening. The results of correct number showed that the similar tendency between two groups. This may indicate that listening to L2 is difficult, as much research has suggested, but if Japanese speakers continue to learn English listening, they can better develop recognition of the spoken word more accurately and the inferring skill from top down information such as grammar and semantics may play a role for listening in the second language.

4pSC19. Cross-linguistic perception of continuous speech: Neural entrainment to the speech amplitude envelope. Jieun Song and Paul Iverson (Speech, Hearing and Phonetic Sci., Univ. College London, Rm. 326, Chandler House, 2 Wakefield St., London WC1N 1PF, United Kingdom, jieun.song@ucl.ac.uk)

Brain oscillations in the auditory cortex become entrained to slow amplitude fluctuations in the speech signal (i.e., amplitude envelope) during speech comprehension. Previous research has suggested that this cortical activity plays an important role in speech perception by aligning neuronal excitability to parts of the incoming speech signal that contain linguistic information such as syllables. However, it remains unclear how cortical entrainment to the speech envelope relates to higher-level linguistic processes during speech comprehension. The aim of the present study was to see if cortical entrainment to the amplitude envelope varies depending on whether or not the listener understands the linguistic content of the speech signal. To this end, the phase-locking between neural oscillations and amplitude envelopes of speech were measured in EEG recordings from listeners with different linguistic backgrounds (i.e., native English and Korean speakers) while they heard continuous speech in three languages (i.e., English, Korean, and Spanish). The results demonstrate that auditory-cortical tracking of the envelope is not affected by whether or not listeners understand the speech, supporting the view that envelope-tracking activity is mainly independent of higher linguistic processes.

4pSC20. Cross-linguistic perception of clearly spoken English tense and lax vowels based on auditory, visual, and auditory-visual information. Keith K. Leung (Linguist, Simon Fraser Univ., Robert C. Brown Hall Bldg., Rm. 9201, 8888 University Dr., Burnaby, BC V5A 1S6, Canada, kwl23@sfu.ca), Charles Redmon (Linguist, The Univ. of Kansas, Lawrence, KS), Yue Wang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Allard Jongman, and Joan Sereno (Linguist, The Univ. of Kansas, Lawrence, KS)

Studies on acoustic and visual characteristics of English tense and lax vowels show consistent enhancement of tensity contrasts in clear speech. However, the degree to which listeners utilize these enhancements in speech perception remains unclear. The present study addresses this issue by testing speech style effects on tense and lax vowel perception by 23 native English and 30 non-native Mandarin-Chinese listeners in audio-only (AO), audio-visual (AV), and visual-only (VO) stimulus modes. English and Chinese listeners showed similar relative differences in performance by mode (VO < AO < AV) and style (plain < clear). However, the two groups differed in the nature of the interaction between tensity, style, and stimulus mode. English listeners showed advantages for clear speech for both tense and lax vowels in all but VO stimuli, whereas Chinese listeners showed a clear speech advantage only for tense vowels, while clear lax vowels showed no improvement in AO and reduced accuracy in AV and VO. While temporal and spectral acoustic cues may coordinate to preserve or improve tense-lax category identity in clear speech, non-native listeners may not be attending to both dimensions. Further, Chinese listeners' greater reliance on visual information may account for their less accurate lax vowel identification.

4pSC21. Functional load of fundamental frequency in the native language predicts learning and use of these cues in second-language speech segmentation. Annie Tremblay (Linguist, Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 60045-3129, atrembla@ku.edu), Mirjam Broersma (Ctr. for Lang. Studies, Radboud Univ., Nijmegen, Netherlands), Caitlin E. Coughlin (Linguist, Univ. of Kansas, Lawrence, KS), and Monica Wagner (Donders Inst. for Brain, Cognition and Behaviour, Ctr. for Cognition, Radboud Univ., Nijmegen, Netherlands)

This study investigates whether second-language (L2) learners make greater use of prosodic cues to word boundaries if these cues have a higher functional load in the native language (L1). It examines the use of fundamental-frequency (F0) rise in the segmentation of French speech by English- and Dutch-speaking L2 learners of French matched in French proficiency and experience (and native French listeners). In both English and Dutch, an F0 rise tends to signal word-initial boundaries in pitch-accented words with initial stress, but English has more vowel reduction than Dutch, and thus full vowels also signal word-initial boundaries in English. By contrast, in French, an F0 rise tends to signal word-final boundaries. Participants completed an eye-tracking experiment: They heard sentences where the monosyllabic target ended or did not end with an F0 rise (duration being held constant), and saw four orthographic words in the display, including the target (*bal* "ball") and a disyllabic competitor (*balcon* "balcony"). Growth curve analyses on participants' fixations revealed that Dutch listeners made earlier and greater use of F0 cues than English (and French) listeners. These results suggest that the functional load of F0 in the L1 predicts the learning and use of these cues in the L2.

4pSC22. The effects of acoustic and lexical factors on perception of accented speech. Christina Y. Tzeng (Emory Univ., 36 Eagle Row, Atlanta, GA 30322, ctzeng@emory.edu), Jessica E.D. Alexander (Centenary College of Louisiana, Shreveport, LA), and Lynne C. Nygaard (Emory Univ., Atlanta, GA)

Foreign-accented speech differs from native pronunciation on a variety of spectral and temporal properties that affect both accentedness and intelligibility. The specific acoustic correlates, however, as well as the role of signal-independent lexical characteristics on accentedness judgments and intelligibility, remain unclear. The current study examined the potential contribution of particular acoustic (F1, F2, duration) and lexical (frequency, neighborhood density) factors to native speakers' ratings of accentedness and assessments of intelligibility for English words spoken by native Spanish speakers. Previous research has found that in native English speech, hard words (low frequency, high neighborhood density) have more dispersed vowel spaces relative to easy words (Pardo *et al.*, 2013). Preliminary findings in the current study indicate the opposite pattern such that foreign-accented easy words exhibited more vowel dispersion than hard words. Further, vowel dispersion measures for easy, but not hard, words predicted accentedness and intelligibility such that as speakers' vowel spaces became more dispersed, the more accented and less intelligible their utterances were rated. Together, these findings suggest that signal-dependent and signal-independent factors have differential effects on the production and perception of accented versus native speech, implying a complex relationship between acoustic and lexical properties that is unique to foreign-accented speech.

4pSC23. Effects of native language on the use of segmental and suprasegmental cues to stress in English word recognition: An eye-tracking study. Katrina Connell (Linguist, Univ. of Kansas, 541 Lilac Ln. Blake Hall, Rm. 427, Lawrence, KS 60045-3129), Simone Hüls (Speech-Language-Hearing: Sci. & Disord., Univ. of Kansas, Lawrence, KS), Maria Teresa Martínez-García (Modern Lang., Fort Hays State Univ., Hays, KS), Zhen Qin, Seulgi Shin (Linguist, Univ. of Kansas, Lawrence, KS), Hanbo Yan (Shanghai Int. Studies Univ., Shanghai, China), and Annie Tremblay (Linguist, Univ. of Kansas, Lawrence, KS, atrembla.illinois@gmail.com)

This study investigates whether the presence of lexical stress in the native language (L1) determines second-language (L2) learners' ability to use stress in L2 lexical access. It focuses on (standard Mandarin) Chinese and (Seoul) Korean listeners' (and native English listeners') use of segmental and suprasegmental cues to stress in English word recognition. Stress placement in English is signaled by segmental (vowel reduction) and suprasegmental (fundamental frequency, duration, and intensity) cues. Chinese has full-full and full-reduced words that differ in stress placement, with segmental and suprasegmental cues signaling stress. By contrast, Korean does not have lexical stress. Participants completed an eye-tracking experiment. They heard stimuli containing a target word with initial stress (*parrot*), and saw four orthographic words in the display, including the target and one of two competitors (stress match: *parish*; stress mismatch: *parade*). The first syllable of the target and stress-mismatch competitor differed in both segmental and suprasegmental information (*parrot-parade*) or only in suprasegmental information (*mystic-mistake*). Growth-curve analyses on fixations revealed that only Chinese and English listeners used stress to recognize English words, confirming L1 effects on the use of stress in L2 lexical access. Furthermore, only English listeners made greater use of stress in the presence of vowel reduction.

4pSC24. The examination of acoustic feature of English obstruent coda by Mandarin and Cantonese speakers. Lu Wanling (Linguist, Univ. of Hong Kong, R4, F5, Block D, Kwan Yick Phase Two, Des Voeux Rd., Hong Kong Island, Hong Kong 999077, Hong Kong, katielu@hku.hk)

Mandarin and Cantonese, though are both Chinese, exhibit differences in L2 English pronunciation, especially in obstruent codas. For the reason that only nasals -n, -m and -ng can be found at coda in Mandarin whereas apart from nasals, unreleased consonants -p -t and -k exits as well in Cantonese, it is reasonable to believe that L1 transfer makes influences on modifying codas for L2 learners. This present study aims at comparing the strategies used by Mandarin and Cantonese speakers in coda modification and resyllabification. By embedding the stimuli in a natural conversation, room-cleaning task, data are elicited and analyzed by Praat. Results indicates that up to 93% of codas are released by Mandarin speakers and 41% of those are unreleased by Cantonese speakers. Further, instead of epenthesizing vocalic elements, Mandarin speakers are found to strongly aspirate the codas with 58%, which contradicts most of the previous studies claiming that epenthesis is the main modification used among Mandarin L2 learners.

4pSC25. Do listeners' expectations about the source of a speaker's accent affect intelligibility? Charlotte Vaughn (Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403-1290, cvaughn@uoregon.edu)

When encountering accented speakers, listeners initially show a processing cost but are able to adapt to the accent within minutes (Bradlow & Bent, 2008; Floccia *et al.*, 2009). However, the extent to which processing a foreign accent (*L2-accented*) is similar to processing a within-language accent (*L1-accented*) is still an open question (cf. Cristia *et al.*, 2012). This project considers whether intelligibility is affected by listeners' expectations about the source of the accent, namely, whether the speaker is believed to be an L1 or an L2 speaker. In this study, native English listeners transcribed English sentences in noise spoken by a bilingual Spanish/English speaker. Expectations about the source of the speaker's accent were manipulated using a matched guise design: Some listeners received no information about the L1 status of the speaker, others were told the speaker was an L1-Spanish speaker, and others were told the speaker was a Hispanic American whose L1 was English (Latino English, e.g., Fought, 2003). Transcription accuracy is compared across conditions to explore whether listeners are sensitive to the kinds of acoustic properties differentiating L1-accented vs. L2-accented speech, and whether listeners' expectations about the source of a speaker's accent affects the intelligibility of that speaker.

4pSC26. The effects of visual and production components in training the perception and production of English consonant contrasts by Mandarin speakers. Janice Wing Sze Wong (Dept. of Commun. Studies, Hong Kong Baptist Univ., 9/F, Commun. and Visual Arts Bldg., 5 Hereford Rd., Kowloon Tong, Hong Kong, Hong Kong, janicewong@hkbu.edu.hk)

Studies have shown the effectiveness of auditory-only training in improving the perception and/or production of non-native contrasts. This study investigated whether providing additional visual and production components are more useful than auditory-only training. The contrasts /ð/-/z/ and /θ/-/s/ were chosen since studies reported these common confusions among Mandarin EFL learners. Forty-five participants were divided into five groups receiving 10 training sessions: auditory-only (A), audio-visual (AV), auditory and explicit production (AP), visual-only (V) or explicit production-only (P). All auditory and visual stimuli were produced by six native General American speakers while the production training involved a native GA speaker giving immediate feedback. Word-level identification and reading pre/post-tests were used to measure their perception and production performance. Robust perceptual improvement of the two consonant pairs was found in A, AV, AP, and P although they did not differ significantly. Their production only improved significantly when they were trained under A, AV, AP, and P, whereas AP and P groups performed significantly better than other groups. These results showed auditory-only training could already improve the perception and production of difficult contrasts, while visual and production components did not make a significant difference in perception improvement. Only direct production training was more beneficial in training participants' production.

4pSC27. Contextual analysis on geminate/singleton identification difficulties for L2 learners of Japanese based on perceptual features. Yanlong Zhang (GITS, Waseda Univ., Rm. 702, Bldg. 66, Nishi Waseda Campus, Okubo 3-14-9, Shinjuku, Tokyo 169-8555, Japan, chouenryu0903@ruri.waseda.jp), Hideharu Nakajima (NTT Media Intelligence Labs., NTT Corp., Tokyo, Japan), Mee Sonu (College of Humanities and Sci., Nihon Univ., Tokyo, Japan), Hiroaki Kato (NICT, Kyoto, Japan), and Yoshinori Sagisaka (GITS / Dept. of Appl. Mathematics. FSE, Waseda Univ., Tokyo, Japan)

It is widely known that Japanese geminate/singleton consonant identification is one of the biggest problems for L2 learners. We have been analyzing identification error characteristics based on their perceptually motivated features. In this presentation, after briefly introducing observed remarkable identification error chracteristics on phonetic context differences with speech rate variations, we have tried to analyze and quantify the difficulties using identification error rates by Japanese beginners of Korean natives. 36 geminate/singleton pairs in 2-5 mora with three speech rates were used for identification experiments. To understand the difficulties quantitatively, objectively measurable acoustic variables such as duration length, loudness and its jumps of the corresponding geminate/singleton consonants were employed to carry out prediction analysis for L2's identification error rates. Correlation between predicted error rates and observed ones has turned out to be around 0.6 with the speech rates where more identification errors existed. Through these analyses, we could have confirmed that perceptually motivated parameters could provide reasonable and effective explanation in contextual differences of identification errors by L2 learners and its quantification. These results suggest the possibility of effective L2 learning based on perceptual difficulties.

4pSC28. Vowel and tone identification of Mandarin Chinese: Comparison between native Chinese and Korean listeners. Hui Zhang, Lilong Xu, Can Xu, Xiaohu Yang, and Hongwei Ding (School of Foreign Lang., Speech, Lang. and Hearing Ctr., Ctr. for Cross-Linguistic Processing and Cognition, 800, Dongchuan Rd., Minhang District, Shanghai 200240, China, zhanghui_Helen@126.com)

Speech in tone languages requires listeners to identify not only phonemes but also tones. As non-tone language users, Korean-native speakers may have difficulty in identifying tones and vowels and processing both tonal and phonemic information at the same time, which are not presented in their mother tongue. The goal of the present study was to investigate the identification of Mandarin Chinese vowels and tones for both native Mandarin Chinese and native Korean speakers with medium and high Chinese proficiency. Results showed that for the identification of Mandarin vowel-plus-tone, vowel, and tone, there was no difference between Chinese- and Korean-native listeners with high Chinese proficiency, whereas Chinese listeners significantly outperformed Korean listeners with medium Chinese proficiency, particularly for tones 1, 3, and 4 and vowel /y/. For Korean listeners, there was no significant correlation between the onset age of Chinese learning and identification scores, while the identification performance significantly correlated with the length of Chinese learning. Moreover, for both Chinese and Korean listeners, identification scores of vowel-plus-tone, tone and vowel are comparable, suggesting that both groups may process dual cues (vowel and tone) as efficiently as individual cues (vowel or tone).

4pSC29. Informational masking of six-talker babble on Mandarin Chinese vowel and tone identification: Comparison between native Chinese and Korean listeners. Can Xu, Hui Zhang, Yuxia Wang, and Hongwei Ding (Speech, Lang. and Hearing Ctr., Ctr. for Cross-linguistic Processing and Cognition, School of Foreign Lang., Shanghai Jiao Tong Univ., 800 Dongchuan Rd., Minhang District, Shanghai, Shanghai 200240, China, 8023xc@sjtu.edu.cn)

The goal of this study was to investigate the informational masking (IM) of Mandarin six-talker babble on Mandarin vowel and tone perception for native Chinese and Korean speakers with high and medium Mandarin proficiency. Mandarin Chinese speech sounds (vowel plus tone) were presented in quiet and two noises: six-talker babble and babble-modulated noise, which was spectrally and temporally matched with babble. The IM was computed as the difference in the identification scores between the two types of noises. Overall, all the four factors (listener group, noise type, vowel-plus-tone category, and SNR) significantly affected participants' identification of Mandarin stimuli. Particularly, in both quiet and noisy conditions, Chinese listeners had comparable performance with Korean listeners with high Mandarin proficiency, while Chinese listeners significantly outperformed Korean listeners with medium Mandarin proficiency. However, no significant difference was found in IM among the listener groups. In conclusion, at the syllabic level, highly-proficient Korean listeners had native-like performance of Mandarin vowel and tone identification in quiet and noise, whereas Korean listeners with medium proficiency had greater difficulty, specifically in noise, likely due to their lower capacity to process phonemic and tonal information, rather than the IM of babble.

4pSC30. Modeling native phonology and non-native speech perception using electroencephalography signals. Daniel McCloy and Adrian K. Lee (Inst. for Learning and Brain Sci., Univ. of Washington, Box 357988, Seattle, WA 98115-7988, drmccloy@uw.edu)

Studies of language learning in adulthood show that learners' native language phonologies shape their non-native perception and production abilities. Nonetheless, listeners are able to learn to perceive new speech sound contrasts given training. The present study attempts to model how foreign consonants are perceived prior to training or second language study. Brain responses were recorded during passive listening using electroencephalography (EEG), using twelve monolingual English-native listeners and isolated consonant-vowel (CV) syllables read by speakers of several languages (Dutch, English, Hungarian, Hindi, Swahili). To model native-language phonology, EEG responses to native (English) syllables were used to train classifiers based on phonological feature labels for the consonants in each syllable. The trained classifiers were then applied to the EEG responses to foreign syllables, and the classifier outputs used to generate confusion probabilities between each pair of foreign and native consonants for each of the four foreign languages. Confusion matrices based on the EEG classifiers compare favorably to confusion matrices based solely on number of phonological feature mismatches. Ongoing work investigates whether optimal phonological features can be learned from the EEG data using clustering algorithms.

4pSC31. Perception of the Japanese moraic-nasal (/N/) by Korean native speakers: Concerning /N/ followed by vowels. Heesun Han (Graduate School of Lang. and Culture, Osaka Univ., 1-8 Machikaneyama, Toyonaka, Osaka 560-0043, Japan, kenkyuhhs@gmail.com)

This study examines the perception of the Japanese Moraic-Nasals /N/ by Korean native speakers. In the case of the /N/ followed by vowels (/N/ + / a, i, u, e, o/), there is a tendency towards pronunciation as various nasal vowels. The actual sound of each /N/ is, however, prone to change according to the speech style. Korean language has three nasal codas (/m, n, ŋ/). The Korean nasal codas are pronounced [m], [n], [ŋ] respectively, regardless of speech styles. Therefore, if Korean native speakers consider Japanese /N/ from the perspective of the Korean nasal coda, it is possible that they will misjudge Japanese /N/. A perception test was conducted using 6 minimal pairs consisting of "/N/ + vowel (eg. /saNin/)" and "vowel + vowel (eg. / saiin/)". The test words were recorded in "common," "emphasized," and "unclear" speech styles. The results showed that native Japanese speakers recognized /N/ in most cases. On the other hand, Korean speakers with no experience in learning Japanese were unable to recognize the Japanese /N/, and perceived /N/ as vowel. Korean Native speakers with advanced Japanese proficiency had results similar to the native Japanese participants. Interestingly, in some cases they showed more accurate judgement than that of the Japanese. This, therefore, suggests that, as language learning progresses, the Japanese /N/ is acquired. Advanced learners, however, tended to pay greater attention to the /N/ sounds rather than the contextual factors on which Japanese were more likely to rely on.

4pSC32. Acoustic analysis of oral and nasal Hindi vowels spoken by native and non-native speakers. Shyam S. Agrawal, Shweta Bansal, Shambhu Sharan, and Minakshi Mahajan (College of Eng., KIIT, Sohna Rd., Near Bhondsi, Gurgaon, Haryana 122102, India, dr.shyamsagrawal@gmail.com)

This paper presents acoustic analysis of Hindi oral and nasal vowels as spoken by the native Hindi speakers and non-native speakers belonging to Punjabi and Nepali linguistic backgrounds. The objective of the study is to determine the measurable acoustic parameters of Hindi vowel sounds which can be used to distinguish oral and nasal vowels by Hindi speakers on one hand and the effect on these parameters due to non-native speakers on the other hand. The database consists of 150 phonetically rich sentences uttered by 10 speakers of each language in a sound treated room. These were analysed using PRATT and Wave Surfer software tools. Five long Hindi vowels (oral as well as nasalized) i.e. /i/, /ɛ/, /a/, /ɔ/ and /u/ were selected for this study and segmented from the sentence utterances. The parameters such as frequencies, amplitude of formants and the pitch and intensity values at the steadiest portion of the vowels were computed. The results show several parameters which distinguishes oral vowel from Nasal vowels as well as the effect of languages of Non-native speakers on Hindi Vowels. Significant observations and the results of the study have been discussed in detail in this paper.

4pSC33. Pitch accents in the Spanish of Japanese-Spanish bilinguals. Tanya Flores (Lang. and Lit., Univ. of Utah, 255 S Central Campus Dr., LNCO 1400, Salt Lake City, UT 84109, Tanya.Flores@utah.edu)

This study examines supra-segmental traits in the Spanish speech productions of thirteen Japanese L1—Spanish L2 speakers living in Valencia, Spain. The analysis presented here will focus on the use of pitch accents and final syllable lengthening. The data in this study shows speakers using Japanese pitch accent patterns in Spanish words, often as the only phonetic indicator of second language speech. The goal of this study is to identify specific Japanese pitch accent patterns used and determine if and how they are systematically transferred into Spanish. Following the autosegmental-metric model (Sp_{ToBI}), pitch accents will be measured using PRAAT. Findings will contribute to the currently limited acoustic research on supra-segmental or prosodic traits of L2 speech in language contact situations.

4pSC34. Bidirectional prosodic effects of code-switching in Spanish-Basque bilinguals. Ann M. Aly (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095, ann.m.aly@ucla.edu)

This study investigates the prosody of code-switching in Spanish-Basque bilinguals, which has not been previously documented. Peninsular Spanish and Basque have several prosodic differences, including peak alignment in pitch accents. Spanish has delayed peaks (Estebas Plana & Prieto (2008)), while Basque has early peaks (Elordieta & Hualde (2015)). The current study investigates whether code-switched words are conditioned by prosodic position and whether the peak alignment and f0 maximum of a pitch accent differ between monolingual context (Spanish only) and code-switched contexts. Bilinguals (aged 21-31) from a Basque-dominant region (Lekeitio, n = 3) and a Spanish-dominant region (Bilbao, n = 4) participated in the present study. The data was coded for the following predictor variables: prosodic position, peak alignment, and f0 maximum in pitch accents. Results reveal that code-switches occur almost equally in phrase-medial and final positions. For peak alignment, Lekeitio speakers produced early peaks in both code-switching contexts (Spanish-Basque, Basque-Spanish), which matches monolingual norms for Lekeitio varieties of Spanish and Basque. However, Spanish-dominant speakers also produced early peaks in both code-switching contexts, despite delayed peak norms for Peninsular Spanish. Finally, preliminary results reveal greater f0 during code-switching contexts for Lekeitio speakers only. Further results and implications will be discussed.

4pSC35. Japanese singleton and geminate stops mispronounced by non-native speakers. Kimiko Yamakawa (Culture and Lang., Shokei Univ., 2-8-1 Musashigaoka-kita, Kikuyo, Kikuchi, Kumamoto 8618538, Japan, jin@shokei-gakuen.ac.jp) and Shigeaki Amano (Aichi Shukutoku Univ., Nagakute, Aichi, Japan)

Non-native Japanese speakers often mispronounce Japanese singleton and geminate stops. However, it has been unclear how durations of speech segments in their mispronunciation differ from those of correct pronunciation by native Japanese speakers. This study analyzed mispronunciation of singleton and geminate stops by non-native speakers (8 English, 12 French, and 10 each of Korean, Taiwanese, Thai, and Vietnamese). Non-native speakers' mispronunciations were identified based on 10 native Japanese speakers' production boundary between singleton and geminate stops. It was revealed that when non-native speakers mispronounce a singleton stop as a geminate stop, stop closure is longer whereas its previous mora is shorter than in native speakers' correct pronunciation. Meanwhile, when non-native speakers mispronounce a geminate stop as a singleton stop, stop closure is shorter whereas its following mora is longer. These findings indicate that non-native speakers' mispronunciations are caused not only by the durations of closure but also by the lengths of the previous/following mora. [This study was supported by JSPS KAKENHI Grants Number JP 25284080, 26370464, 15H03207, and 16K13221. It was also supported by a special-research grant of Aichi Shukutoku University in 2015-2016.]

4pSC36. Production and reproduction of Japanese pitch-accent by German, Italian, Vietnam, and Swahili speakers. Ryoko Hayashi (Graduate School of Intercultural Studies, Kobe Univ., Nada-ku Tsurukabuto 1-2-1, Kobe, Hyogo 6578501, Japan, rhayashi@kobe-u.ac.jp), Kazuhiro Isomura (The Japan Foundation Japanese-Lang. Inst., Tokyo, Japan), Makiko Matsuda (Kanazawa Univ., Kanazawa, Japan), Natsuya Yoshida (Hokkaido Bunkyo Univ., Sapporo, Japan), and Motoko Ueyama (Univ. of Bologna, Forli', Italy)

Japanese pitch accent is lexically distinctive. JFL learners should know the position of the accent to pronounce the words correctly, which is said to be very difficult. In our experiment, we investigate the effectiveness of showing the accent mark to help learners' pronunciations. We compared the ability of perception, production and reproduction of pitch accent by JFLs of four different languages. 42 Japanese words were presented on a PC screen and 30 subjects participated in three tasks of the experiments. Task 1 was to pronounce the words without any suggestion, task 2 was to imitate the model pronunciation with seeing the accent marks. Task 3 was to pronounce by just seeing the accent marks. The results of task 1 show the percentage of correct accent position was only around 40%, but it rose up to around 90 % in task 2 and fell again in task 3 but still significantly higher than in task 1. The results of task 1 showed different patterns of phonetic interference, task 2 showed that the subjects could easily perceive the position of accent and imitate the pitch pattern. Task 3 implied the effectiveness of showing the accent mark.

4pSC37. Tonal realization of English intonation and register by Japanese learners of English as a foreign language compared to English speakers. Toshiko Isei-Jaakkola (College of Humanities, Chubu Univ., Matsumoto 1,200, Kasugai, Aichi 487-8501, Japan, tiseij@isc.chubu.ac.jp) and Keiko Ochi (National Inst. of Informatics, Chiyoda-ku, Tokyo, Japan)

Theoretically, there are four levels of tone in English intonation, whereas in standard Japanese, phonologically, there are two levels (high and low) and phonetically, there are three levels of in lexical tone. The Japanese said to be a mora-timed and pitch-accented language and English to be a stress-timed. The Japanese language sounds monotonous. Further, English speakers (EL1) seem to have a wider register than Japanese learners of English as a foreign language (JEFL). There is only one kind of interrogative sentence in Japanese, while there are two kinds in English. Moreover, there is a difference in the position of the nucleus between Japanese and English. This study aims to examine whether JEFL can realize the four levels of tone in English, and the degree of difference in register between JEFL and EL1 when producing a sample English intonation pattern. We conducted experiments to investigate JEFL's realization of English intonation in one word-sentences, affirmative sentences, and interrogative sentences with rising,

falling, and rising-falling tones. We found that JEFL could produce intonational contours resembling the model English sentences but with insufficient nuclei (weak intensity), and that EL1 have a wider register than JEFL in both minimum F0 and maximum F0 as presumed.

4pSC38. Speech rhythm characteristics of L2 English produced by Japanese. Saya Kawase, Jeesun Kim, and Chris Davis (MARCS Inst., Western Sydney Univ., Locked Bag 1797, Penrith, NSW 2751, Australia, S.Kawase@westernsydney.edu.au)

This study investigated the durational rhythm characteristics of L2 English produced by L1 Japanese talkers with a particular focus on the interaction between L1 and L2 speech rhythm and the role of L2 experience. More specifically, for English sentences (N = 40) spoken by native Japanese (N = 10) and native Australian English (N = 10) talkers, we examined (1) mean consonant and vowel durations and durational variability and (2) vowel and consonant timing patterns within consonant clusters. Half of the Japanese talkers had more experience in L2 English. L2 productions had longer consonant and vowel duration and less variability compared to native English ones. The degree of L2 experience played a role: inexperienced L2 talkers produced less variable vowel durations than the experienced ones. The analyses of speech timing patterns in consonant cluster productions showed that inexperienced Japanese talkers produced significantly shorter second consonants in consonant clusters compared to native English productions, possibly compensating for the difficulty in producing nonnative consonant clusters. Additional analyses of the pattern of consonant cluster production by experienced L2 talkers will also be reported.

4pSC39. Types of errors in English prosody made by native speakers of Korean. Heeju Lee and Sun-Ah Jun (UCLA, 290 Royce Hall, Box 951540, Los Angeles, CA 90095-1540, heejulee@ucla.edu)

This study investigates the types of errors that are made by speakers of L1 Korean in the production of L2 English prosody. English and Korean differ in prosodic and rhythmic structure and in the ways word prominence and semantic/pragmatic meaning are marked. In English, a prominent word is marked by a pitch accent on its stressed syllable, and semantic/pragmatic information is delivered by pitch accent types (e.g., H*, L + H*) and boundary tones. In Korean, a language without lexical stress, a word's prominence is marked by its location at the beginning of a phrase, and semantic/pragmatic information is delivered by prosodic phrasing and boundary tones. Data were selected from recordings of the UCLA TOP (Test of Oral Proficiency) exam taken by Korean students. The present study analyzes the part where students (twelve) present information from the same syllabus in front of raters and questioners acting as students. Control data from three native English speakers was collected in the same format. Prominence, phrasing, and intonation patterns were marked following English ToBI conventions. Preliminary results indicate that the types of errors can be distinguished in terms of prosodic phrasing, location and types of pitch accent, and realization of pitch accents and stressed/unstressed syllables.

4pSC40. Oral gestural reduction of English nasal coda produced by Mandarin English-as-a-foreign-language learners. Yadong Liu (Linguist, The Univ. of Hong Kong, Run Run Shaw Tower, Hong Kong, Hong Kong, yadong@connect.hku.hk)

Mandarin post-vocalic nasal production studies focus on vowel-to-nasal formant transitions (e.g. Jian and Le [(2014), Information Technology Journal, **13**(11), 1793-1799] show the vowel-to-nasal boundary was absent in the context of VN.V). However, it remains unclear whether this feature of Mandarin speakers' L1 extends to their L2 productions. The present study investigates five Mandarin EFL learners' articulation of English post-vocalic nasal codas /n, ŋ/ and homorganic oral codas /d, g/ in monosyllabic words in pre-vocalic contexts. Articulatory data from ultrasound-imaging were analyzed for lingual gesture achievement (in terms of aperture), where a reduced oral constriction corresponds to absence of a stop. The results show reduction of the oral constriction of both /n/ and /ŋ/ relative to their oral counterparts. Further, the reduction varied by place, with greater reduction in /ŋ/ compared to /n/. Additionally, the height of the preceding vowel affects oral constriction reduction of both nasals, with low vowels eliciting more reduction than high vowels. This research studies nasal provides evidence of oral constriction reduction during Mandarin EFL learners' English nasal coda production from an articulatory perceptive, consistent with the L1 nasal closure reduction shown in Jian and Le (2014).

4pSC41. Adaptation to foreign-accented speech with decreased spectral resolution. Michelle R. Kapolowicz, Vahid Montazeri, James F. Kuang, and Peter F. Assmann (Behavioral and Brain Sci., The Univ. of Texas at Dallas, 800 West Campbell GR 4.1, Richardson, TX 75080, mrk092020@utdallas.edu)

Listening to foreign-accented speech requires greater effort, and intelligibility is lower than for native speech. We have recently shown that speech processed by a vocoder to simulate reduced spectral resolution imposes a substantial additional cost to intelligibility. The present study investigated perceptual adaptation to foreign-accented speech in conditions with reduced spectral resolution using a 9-channel tone vocoder compared to unprocessed speech. Preliminary results suggest that listeners can quickly adapt to foreign-accented speech with decreased spectral resolution. Native American English listeners achieved performance levels similar to those obtained with initial exposure to the same unprocessed speech from multiple talkers of Mandarin-accented English. However, with increased exposure, performance did not improve to the same degree found for unprocessed foreign-accented speech. Results indicate that the adaptation process is negatively impacted by the interaction of foreign accent and reduced spectral resolution, and suggests that special consideration should be given to the difficulties experienced by listeners with cochlear implants (whose devices provide limited spectral resolution) when they communicate with speakers who have a foreign accent.

4pSC42. Japanese English learners use a variety of articulation strategies to produce English liquids. Jeffrey Moore (Sophia Univ., Chiyoda-ku, Kioi-cho, 6-2, Tokyo, Tokyo 102-0094, Japan, jeffmoore@sophia.ac.jp), Jason Shaw (Yale Univ., New Haven, CT), Shigeto Kawahara (Keio Univ., Tokyo, Japan), and Takayuki Arai (Sophia Univ., Tokyo, Japan)

This study examines the production of English liquids /r/ and /l/ by Japanese speakers. Six speakers of varying levels of proficiency were recorded using an NDI Wave Electromagnetic Articulograph. They produced words containing /r/ and /l/ in initial singletons and initial clusters. Two native speakers of English were also recorded for comparison. The resulting articulatory data are analyzed for spatial targets, gestural timing, and the amount of variation between utterances. Results indicate a variety of strategies employed by speakers to achieve the non-native targets; however, more native-like, though distinctly Japanese-accented production, among advanced speakers was found. Lower-proficiency and intermediate speakers showed relatively large amounts of within-speaker variation, indicating the ongoing development of articulation strategies. These findings suggest that the acquisition of foreign language sounds is not merely a straight path from L1 transfer to target-like pronunciation, but a complex process involving many paths, hypothesis testing, and experimentation to find a suitable articulation.

4pSC43. Articulatory variations of Mandarin retroflex consonants produced by second language speakers: An electromagnetic articulograph study. Haruka Saito (School of Commun. Sci. and Disord., McGill Univ., 522 Ave. des Pins, Rm. #5, Montreal, QC H2W 1S6, Canada, haruka.saito2@mail.mcgill.ca)

This study investigated articulatory variations in tongue positions and shapes during the production of Mandarin post-alveolar retroflex consonants produced by speakers with varying Mandarin proficiency (native Mandarin speakers, Japanese L2 speakers of Mandarin, and Japanese monolinguals with no knowledge of Mandarin). We aimed to examine (1) whether there are articulatory variations for Mandarin retroflex consonants and (2) if preferred variations differ across groups. Speakers either read aloud or imitated the consonants after hearing them and their tongue positions and shapes were measured by electromagnetic articulography (WAVE, NDI). Results showed that there are multiple articulatory variations for Mandarin retroflex consonants. Native Mandarin speakers produced a concave or convex

tongue shape; all Japanese L2 speakers of Mandarin produced a convex tongue shape. In contrast, the majority of Japanese monolinguals imitated the sounds with an entirely different tongue position: bunching their tongues in the middle, which somewhat resembled the "bunched" rhotic in American English. Despite these articulatory variations, productions by most L2 speakers and several monolinguals were successfully identified as retroflex consonants by native Mandarin listeners. These results suggest that L2 speakers may prefer certain articulatory variations and the preference may change depending on proficiency.

4pSC44. A limited role of hand gestures and head nods in native English speakers' production of Mandarin tones. Yukari Hirata (Ctr. for Lang. and Brain, Colgate Univ., 13 Oak Dr., Hamilton, NY 13346, yhirata@colgate.edu), Annie Zheng (Psych., Univ. of Texas at Austin, Austin, TX), and Spencer D. Kelly (Ctr. for Lang. and Brain, Colgate Univ., Hamilton, NY)

Theories of embodied cognition have emphasized the importance of the whole body in human communication (Lakoff & Johnson, 1999), and numerous empirical studies have shown that bodily actions affect multiple aspects of speech production and perception. This study examined the roles that hand and head gestures play in production of Mandarin tones by native English speakers. Twenty-four native English speakers were asked to imitate the four Mandarin tones from a model across three conditions: (1) audio-only, (2) audio and hand gestures, and (3) audio and head nods. For (2) and (3), participants watched videos in which a native Mandarin speaker visually depicted the tones with her hand and head, respectively. The participants' production was then presented to seven native Mandarin listeners in an identification test to determine percent correctly produced. A 3 (conditions) x 4 (tones) ANOVA revealed a significant main effect of tone ($1 > 3$, $4 > 2$) but no main effect or interaction of condition, indicating that hand and head gestures did not help learners accurately produce Mandarin tones. The results corroborate earlier acoustic analysis of the production data, confirming a limited role of hand and head gestures in phonetic and phonological aspects of L2 learning.

4pSC45. Collocational patterns of disfluencies in native and non-native speech: Evidence from the Wildcat Corpus. Misaki Kato, Tyler Kendall, and Melissa Baese-Berk (Linguist, Univ. of Oregon, 270 Straub Hall, 1290 University of Oregon, Eugene, OR 97403, misaki@uoregon.edu)

While disfluencies are generally considered a natural part of spontaneous speech, patterns of disfluencies in non-native (NN) speech could contribute to making that speech sound less proficient or less "fluent". NN speech has been described as having more frequent and longer pauses, pauses at within-clause boundaries, shorter mean length of runs, and slower speech rate, compared to native (N) speech (e.g., Riggenbach, 1991; Trofimovich & Baker, 2006). However, in order to understand what makes NN disfluency patterns unique, it is important to examine such phenomena locally by examining the contexts of different types of disfluency. In the present study, we describe what features collocate with disfluencies in spontaneous English dialog. We examine speech from the Wildcat Corpus (Van Engen *et al*., 2010) which includes English dialogue produced by various pairings of N and NN speakers (N-N, N-NN, or NN-NN). Specifically, we extract turn-internal disfluencies of several types (e.g., silent pause, filled pause, repetition), and examine lexical (e.g., part-of-speech) and phonetic (e.g., vowel length) features of speech surrounding these disfluencies. Further, we examine whether these disfluency patterns change depending on speaker pairings. Differences between N and NN disfluency patterns are discussed in terms of their potential sources (e.g., speaker- vs. listener-oriented).

4pSC46. Transfer of L1 tone knowledge to L2 vowel quantity contrasts as a secondary cue: The case of Cantonese learners of Japanese. Chun Yin Liu (The Univ. of Hong Kong, Pok Fu Lam Rd., Hong Kong, Hong Kong, yin2013@hku.hk) and Yukari Hirata (Ctr. for Lang. and Brain, Colgate Univ., Hamilton, NY)

Distinguishing vowel quantities in Japanese is problematic for Cantonese-speaking learners. Vowel length is lexically contrastive in Japanese but not in Cantonese. Apart from duration, fundamental frequency (F0) patterns are secondary cues to vowel length distinction. As tone is lexically contrastive in Cantonese (particularly rising vs. high level), learners would likely be able to transfer this phonological knowledge to help themselves acquire vowel length contrast in Japanese. A perception experiment was conducted in which learners listened to Japanese disyllabic words. The first syllable of which was lengthened according to the within-word vowel duration ratio in eight steps. A level or dynamic (rising or falling) F0 pattern was then imposed on the first vowel. Similar to other L2 learners, Cantonese-speaking learners did not perceive the long vs. short contrast categorically, but gradiently. While both dynamic F0 patterns signal a long vowel, they only exploited the rising pattern to recognise a long vowel. Our study suggests that learners have assimilated the falling contour to high level, both of which belong to the same tonal category in Cantonese. The theoretical implications of these results are discussed.

4pSC47. Native listeners' evaluation of natural and synthesized prosody in Mandarin of American learners. Ying Chen, Xueqin Zhao, and Li Liu (School of Foreign Studies, Nanjing Univ. of Sci. and Technol., 200 Xiaolingwei St., Nanjing, Jiangsu 210094, China, ychen@njust.edu.cn)

Compared to duration and intensity, F0 was found the most difficult acoustic parameter to acquire in L2 prosody, especially post-focus compression of F0 (Chen, 2016). This study examined three groups' Mandarin production of prosodic focus: native Beijing speakers, early American learners and late American learners. PENTAtrainer2 (Xu & Prom-on, 2014), a data-driven system for prosody analysis and synthesis, was used to model and synthesize F0 contours based on speaker groups and layered annotations of communicative functions: lexical, sentential and focal. Native Mandarin speakers were recruited to identify focus status (neutral, initial, medial, or final focus) and rate the naturalness (1-5 scale) of original and synthesized speech. Results reveal that natural speech was recognized and rated better than synthesized speech, early learners' speech better than late learners' speech, focused sentences better than no-focus sentences, and initial focus and medial focus better than final focus. Tones of focused words interacted with focus status of the sentence and speaker group. Future work will involve pairwise shape comparisons, root-mean-square error, and Pearson's correlation coefficient comparing between natural and synthesized F0 contours. [This work was supported by the National Science Foundation of China 61573187 and Fundamental Research Funds for the Central Universities in China NJUSTWGY14001.]

4pSC48. Articulatory habit versus adaptive flexibility in L2 phone learning. Matthew Faytak (Linguist, Univ. of California, Berkeley, 2632 San Pablo Ave., Apt. A, Berkeley, CA 94702, mf@berkeley.edu)

Learning a motor task is thought to be facilitated when it can be achieved within a set of thoroughly-learned motor patterns, here termed *articulatory habit*. Learners of novel phonological categories in an L2 may first attempt to produce a novel sound by working over the possibility space of acoustic outputs defined by well-known articulatory dimensions. A side effect of this neuromuscular habituation to L1 may be a limit to the adaptive flexibility of articulatory resources to the L2. To evaluate the role of L1 habit in individuals' L2 category productions, 18 L1 English learners of French repeated French stimuli from a model talker while ultrasound and video of lip movement were recorded. Stimuli include the front rounded vowels /y ø/, which are outside of the subjects' articulatory habit in inter-articulator coordination and the particulars of tongue position. A second phase of the study captured L1 English articulatory habit by eliciting a range of English words and recording articulator activity in the same way. Principal components analysis was performed on raw image data of both phases in order to gauge L1-L2 articulatory similarity and to highlight any dimensions not exploited in L1 but observed in L2 or vice-versa.

4pSC49. Age and consonantal context effects on second language speech perception of French vowels. Andrew Lee (McGill Univ., 3700 McTavish St., Montreal, QC H3A 1Y2, Canada, andrew.lee@mcgill.ca)

This study investigated the effects of age of acquisition (AOA) and consonantal contexts on second language (L2) speech perception of French

vowel pairs /i/-/y/ and /y/-/u/. A total of 60 Korean learners of French participated in either an Early-AOA group (under 16 years), a Mid-AOA group (16-26 years), or a Late-AOA group (over 26 years). To measure their perceptual accuracy, AX categorical discrimination tasks were employed in which the two target pairs, in addition to a control pair /i/-/u/, were provided with three different consonants (i.e., /p/, /t/, and /k/) in a consonantal-vowel-consonantal structure. Overall, the Korean participants had more difficulty discriminating the pair /i/-/y/ than the pair /y/-/u/. In particular, the Korean participants revealed more difficulty with the former pair in the /p/ context, whereas they showed more difficulty with the latter pair in the /p/ and /t/ contexts. While those in the Early-AOA group significantly outperformed those in the Mid- and Late-AOA groups, those in the former group perceived the two target pairs in a way that was less affected by the consonantal contexts. Accordingly, this paper will conclude by highlighting the importance of AOA and consonantal contexts in L2 speech learning.

4pSC50. Effects of language bias and proficiency on cross-language activation: Evidence from eye-tracking. Maria Teresa Martinez-Garcia (Modern Lang., Fort Hays State Univ., 600 Park St., Hays, KS 67601-4099, mtmg87@gmail.com) and Annie Tremblay (Linguist, Univ. of Kansas, Lawrence, KS)

Language bias and proficiency have been proposed to modulate cross-language activation, but it is unclear how they operate and whether they interact. This study sheds further light on this by investigating whether stress differences between Spanish-English cognates (*material*, final-syllable stress in Spanish) affect how native-English second-language-Spanish bilinguals recognize Spanish words (*materia* "subject/matter," second-syllable stress in Spanish). In a Spanish-English eye-tracking experiment, participants heard trisyllabic Spanish targets with second-syllable stress (*materia*) and saw four orthographic words, including the target and a Spanish-English cognate competitor. Cross-language activation was examined by manipulating the stress of the cognate in English; English cognates with the same stress as the Spanish target (target: *materia*, competitor: *material*) were predicted to cause more cross-language interference than English cognates with a different stress (target: *litera* "bunk bed," competitor: *literal*). Participants were assigned to a Spanish-bias condition, with 20% of English (filler) items, or an English-bias condition, with 65% of English (filler) items. Spanish proficiency was assessed with a cloze test and LexTALE. Results indicate that English cognates with the same stress as the Spanish target interfered with the recognition of the Spanish target only in the English-bias condition, and bilinguals better controlled this cross-language interference with increasing Spanish proficiency.

4pSC51. The novel Ventriloquist paradigm: Studying L2 phonetic learning in dialogue with experimental control over phonetic detail. Mirjam Broersma (Ctr. for Lang. Studies, Radboud Univ., P.O. Box 9103, Nijmegen 6500 HD, Netherlands, m.broersma@let.ru.nl)

The novel Ventriloquist paradigm enables the study of phonetic accommodation and perceptual learning in dialogue, while allowing full control over the phonetic detail of the input that participants are exposed to. It is being developed to investigate L2 speech learning for production and perception in an ecologically valid yet maximally controlled way. Participants take part in a dialog which they believe to be genuine; in fact, however, their (real-life) interlocutor is a confederate whose speech is not just "scripted" (as in the confederate scripting task), but fully prerecorded. This guarantees control over all characteristics of the speech input, e.g., precluding that a confederate accommodates his/her pronunciation to the participant. The set-up is fully tuned to upholding the illusion that the confederate is actually speaking with the participant. The confederate sits opposite the participant, face briefly hidden when he/she "speaks." Participants hear the prerecorded speech over closed headphones. In addition to the standard input, to facilitate a smooth flow of the conversation, the confederate can play prerecorded non-verbal interaction markers and stop-gap replies to any unanticipated remarks or questions from the participant. The new paradigm thus reconciles ecological validity with experimental control for the study of (L2) phonetic processing in dialog.

4pSC52. Non-native perception and production of lexical tones before and after high-variability training. Yan Chen (Linguist Dept., Univ. of Arizona, Tucson, AZ 85721, yanchen@email.arizona.edu)

This study investigates non-native perception and production of lexical tones. 26 English speakers received 3 hours of high-variability AX training to perceive tonal contrasts in Cantonese. Half of them were trained with additional iconic tone marks (AV) and half of them were not (AO). Pre-test and post-test were conducted and both of them consisted of high-variability AXB followed by delayed repetition using the stimuli from AXB. Preliminary results showed that T2 (high-rising) vs. T5 (low-rising) and T3 (mid-level) vs. T6 (low-level) were the most difficult pairs in AXB, with about 70% correct for T2-T5 and slightly above chance performance for T3-T6 in post-test. AV listeners did not achieve significantly higher accuracy than AO listeners. In delayed repetition, both groups produced distinct contours for T2 and T5 and for T3 and T6 even in pre-test, which did not differ significantly from post-test. Exposure to pre-test AXB might have facilitated talker normalization and tone production in the delayed repetition task. In addition, AV subjects had slightly more semitone difference in the second half of rhyme (offset—50% of rhyme duration) for T2 in post-test than AO subjects, but this effect of iconic visual information is limited.

4pSC53. Neural changes accompanying overnight nonnative phonetic learning. F. S. Earle and Emily B. Myers (Dept. of Speech, Lang. and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT 06268, frances.earle@uconn.edu)

Sleep is important in the consolidation of learning, including the perceptual learning of non-native speech sounds. Previous work investigating the role of post-training sleep on nonnative phonetic learning has demonstrated that sleep significantly improves identification for a trained talker in the absence of further training. In addition, sleep appears to facilitate generalization of learning to non-native tokens produced by an untrained talker. We investigated the neural correlates of these behavioral observations using fMRI. Participants were trained in the Hindi dental-retroflex contrast in the evening on Day 1. Immediately following training, participants performed the category identification task in the scanner, and then again 12 hours later. Sleep duration was monitored via wrist Actigraphs. On Day 1, significant differences in activation for the Trained vs. Untrained Talker were identified in the supramaginal gyrus and adjacent parietal areas, whereas on Day 2, activation for both talkers was observed to move into the temporal lobes bilaterally. After sleep, and in the absence of further training, decreases in activation were observed in right superior temporal gyrus, right precentral gyrus, and left prefrontal cortex. Based on these findings, we argue that newly acquired nonnative sounds are processed in classic (i.e., more native-like) speech regions following sleep.

4pSC54. Effects of perceptual training on the perception of Korean boundary tones by Chinese learners. Ho-Young Lee, Hyosung Hwang (Linguist, Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul 08826, South Korea, hshwang415@gmail.com), and Eunkyung Yoon (Daegu Cyber Univ., Daegu, South Korea)

Previous studies have proven that High Variability Phonetic Training (HVPT) is an effective method for training non-native listeners to discriminate L2 phoneme contrasts or prosodic contrasts, such as Chinese tone and Japanese segment length. However, it still remains uninvestigated whether this training method can be successfully applied to the perceptual training of nondistinctive L2 intonation. This study aims to prove that the HVPT method is also effective in training Chinese learners of Korean to distinguish Korean boundary tones. Korean boundary tones have three important grammatical and pragmatic functions: distinction of declarative and interrogative sentences, distinction of wh- and yes-no questions, and distinction of ambiguous meanings of some terminal endings. Our experiment is designed to train Chinese learners to discriminate these three functions using the HVPT method. 40 Chinese learners participated in 10 online sessions. They were exposed to 276 sentences delivered by different speakers each session. The subjects showed significant improvements in discriminating all three intonational functions in the posttest. Lower proficiency learners showed greater improvements. A notable change occurred in the development of categorical perception. The category boundaries between declarative and interrogative sentences shifted closer to the native speakers' after the training.

4pSC55. Relationship between learners' attention and perception training effects of a foreign language: Based on questionnaire study to young learners.
Yuko Ikuma (English Education, Osaka Kyoiku Univ., 4-1-1-801, Bingo-cho, Nada-ward, Kobe, Hyogo 657-0037, Japan, yyikuma13@gmail.com)

Many previous practices studied learners' attitude in the field of the second language (L2) research. They observed the effectiveness of the improvement of learners' motivation or instruction for enhancing learners' self-efficacy. From the author's series of the preceding training research targeting at over one thousand young students who utilized self-study-typed computer-assisted language learning (CALL) system, the author has found that the continuous perceptual training be important in order to maintain the ability. This paper focused on the Japanese-speaking young learner's attitude and its progress towards L2 speech perception training mainly based on the questionnaire response after longitudinal training period using CALL materials. The result showed that the accuracy rates for phoneme and rhythm perception tasks improved; that tendency was observed slightly strong on the students who were interested in "phoneme discrimination" or "conversation" from the questionnaire response. It was also demonstrated that higher proficiency students showed interest in broader perspective like "conversation," while lower students in rather narrower like "discrimination," or "sound." These results suggest that learners who got attention toward phoneme might improve their perception, however, to promote their proficiency, they need some instruction to have attention to wider perspective of speech. [Work supported by JSPS 26780498.]

4pSC56. Preliminary analysis of training non-native sounds in noise.
Hinako Masuda (Faculty of Sci. and Technol., Seikei Univ., 3-3-1 Kichijoji Kitamachi, Tokyo, Musashino-shi 180-8633, Japan, h-masuda@st.seikei.ac.jp)

Past research has repeatedly shown that perceiving speech in non-laboratory environments is challenging for both native and non-native listeners, even if they are able to perform well in laboratory (quiet) environment. However, whether the challenge of non-native listeners' perception of non-native sounds in noise can be overcome with training has not been fully investigated. The present study reports preliminary analysis of training Japanese learners of English, focusing on the identification of English voiceless fricatives and approximants in multispeaker babble noise, as Japanese learners of English are well-known for having difficulty in accurate perception of such sounds (e.g., Lambacher *et al.*, 2001; Masuda 2016 among others), especially in noisy listening environments. The experiment consists of three sessions: 1) pre-test, 2) training sessions in noise, and 3) post-test. Identification rates of 1) pre- and 3) post-tests will be compared to investigate the effect of 2) training in noise. The effect of English learning experience will also be discussed.

4pSC57. Noninvasive brain stimulation to facilitate foreign language speech-sound learning in low-aptitude learners.
Tyler K. Perrachione, Sara C. Dougherty, Ja Young Choi, and Elly R. Hu (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, tkp@bu.edu)

Native English speakers with poor pitch-contour identification abilities also have a low aptitude for learning lexical tones, as in Mandarin Chinese. In a four-day training paradigm, native English-speaking young adults with low pitch-contour perception abilities learned a vocabulary incorporating minimal contrasts based on lexical tones (with level, rising, and falling contours), in which they matched spoken words to photographs of objects. Participants were assigned to two groups: The "stimulation" group received anodal transcranial direct current stimulation (tDCS) over left inferior frontal lobe at 2mA during training; the "sham" group underwent an identical set-up procedure each session, but did not receive brain stimulation during training. Participants' learning trajectories, learning attainment, and ability to generalize word recognition to novel talkers were compared between groups and to previously published training data from both low- and high-aptitude learners. Active brain stimulation improved learning for some, but not all, low-aptitude learners. Some low-aptitude learners who received brain stimulation performed as well as high-aptitude learners in previous studies. Low-aptitude learners who received sham stimulation learned no better than in previous studies. These results suggest noninvasive brain stimulation may help recover speech-sound learning abilities in listeners with low pre-training perceptual aptitude.

4pSC58. Lexical training using accented speech improves non-native contrast discrimination.
Navin Viswanathan and Annie J. Olmstead (Speech-Language-Hearing, Univ. of Kansas, 1000 Sunnyside Ave., Lawrence, KS 66045, navin@ku.edu)

Native American-English (AE) listeners have difficulty distinguishing between dental and retroflex stop consonants. For example, when asked to classify voiceless retroflex and dental stops, AE listeners typically label both items as the alveolar /t/. While AE lacks the dental-retroflex distinction, varieties of Indian English (IE) use this contrast to instantiate the AE /t/-/ð/ distinction (*tanks* vs. *thanks*). Specifically, the dental /t̪/ is used in instances where /ð/ would be used in AE while the retroflex /ʈ/ is used in place of /t/. This situation provides us with the opportunity to present the dental-retroflex difference in naturally produced tokens of English. In the current set of experiments, we examine the effect of exposure to IE tokens on AE listeners' discrimination of /t̪/ and /ʈ/. Participants transcribed English words spoken by an IE speaker. The critical items were words that began with either /t/ (e.g., *take*), /ð/ (e.g., *thorough*), or were /t/-/ð/ minimal pairs (*tanks* vs. *thanks*). An AXB discrimination task performed before and after the transcription showed that AE speakers improved in their discrimination of /t̪/ and /ʈ/ after transcription. Subsequent studies examine this learning effect to investigate the role of specific lexical contexts and of speaker-specific information.

4pSC59. Sleep facilitates talker generalization of accent adaptation.
Xin Xie, F. Sayako Earle, and Emily B. Myers (Dept. of Speech, Lang. and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Storrs, CT 06269, xin.xie@uconn.edu)

Lexically-guided phonetic retuning helps listeners adapt to non-canonical productions produced by a given speaker (e.g., a foreign-accented speaker). Our previous work shows that when tested immediately after training on one accented talker, listeners generalized phonetic retuning only to accented talkers that were acoustically similar to the trained talker. In the present study, we tested whether sleep-mediated consolidation promotes talker generalization to a novel talker who is *not* acoustically similar to a trained talker in their phonetic realizations. Focusing on word-final alveolar stops, we examined native-English listeners' 1) adaptation to a trained Mandarin-accented talker and 2) generalization to another untrained Mandarin talker. Participants were tested in a categorization task immediately after training in either the morning or evening and again after twelve hours. Morning and evening groups had comparable performance for the trained talker during the second test, suggesting maintenance of talker-specific learning. Importantly, only evening-trained participants, who had slept before the second test, gained an advantage for the novel talker, whereas morning-trained participants did not improve over the 12 hours in wake state. It is argued that sleep helped listeners to abstract away from specific acoustic properties of the trained talker and thereby facilitated generalization to the novel accented talker.

4pSC60. The effect of training of poetry recitation on English rhythm of Cantonese native speakers.
Angel L. Chui (Dept. of Linguist, The Univ. of Hong Kong, Hong Kong, Hong Kong, u3518907@hku.hk)

English rhythm, although reported a hurdle to many ESL students, remains one of the least taught aspects of pronunciation. Despite the key role rhythm plays in facilitating comprehension and reducing foreign accentedness, systematic training in the ESL classroom is rare. To promote the teaching of rhythm in the local curriculum, this study explores the effect of poetry recitation training on advanced learners' acquisition of English rhythm. Two groups of advanced Cantonese-speaking English learners were recorded reading poems of different foot patterns as well as non-poetic materials (BKB sentences and a fable), each on a separate occasion. As reflected from rhythmic metrics, the group trained in recitation manifest nativeness better despite the similar proficiency level of the two. I, therefore, argue that poem recitation is an effective means to help ESL learners acquire native-like rhythm in English.

4pSC61. Development of a visual app for improving learner's pronunciation with ultrasound and the speech accent archive. Kyori Suzuki and Ian Wilson (Univ. of Aizu, Tturuga, Ikki-machi, Aizuwakamatsu, Fukushima 965-8580, Japan, s1210136@gmail.com)

Although many language learners desire to improve their pronunciation of a foreign language, there are not many apps to help them do so. Most commercial apps for pronunciation evaluation and training focus on only the acoustic signal. However, few systems give visual movements of native speakers' lips, tongue, and jaw. In this paper, we describe the ongoing development of and app that is programmed in Swift for iPhone. The app incorporates and links together different kinds of phonetic data for the pronunciation learner—for example, recorded frontal and side videos of a native speaker's face during pronunciation with an ultrasound movie of the tongue moving in the mouth overlaid. The training text is a paragraph from the Speech Accent Archive. The initial version of this app has two systems. First, listening to English sentences by a native speaker and checking tongue movement with ultrasound. Second, this app has buttons that play from user-chosen words and it also plays in slow motion. The method of these systems may help people to learn a second language more easily and accurately by letting them shadow the audiovisual speech of a native speaker. This app will be demonstrated in front of the poster.

4pSC62. Analysis of the effects on pronunciation of training by using song or native speech. Saori Nemoto, Ian Wilson, and Jeremy Perkins (Univ. of Aizu, Tsuruga, Ikki-machi, Aizuwakamatsu, Fukushima 965-8580, Japan, hy3148tay@gmail.com)

This research is an investigation of whether Japanese speakers' English pronunciation improves more after training on sung or spoken speech. The stimulus was a 14-word sentence taken from one English song's lyrics, and it had some words that are difficult to pronounce for most Japanese English learners. Thirty Japanese learners of English were recorded before training. Then, half of them trained by listening to the English song and singing it, and the other half trained by listening to a native speaker speaking the lyrics. Each group was allowed to train individually for 10 minutes, and then were recorded again. Then, 15 native or near-native English speakers at an American university and 100 native English speakers from Amazon Mechanical Turk evaluated those randomly-presented recordings. Listeners gave points for various phrases' pronunciation, the overall accent and the overall intonation. As a result, we found out that training by using the music condition resulted in generally worse results than the regular speech training. In addition, perhaps surprisingly, intonation of the whole sentence had an additional significant negative effect following music training. These results seem to show that training by using regular speech is more effective for English learners than training by using songs.

4pSC63. Developing ultrasound overlay videos for SENĆOŦEN language learners. Heather Bliss, Sonya Bird (Linguist, Univ. of Victoria, PO Box 1700 STN CSC, Victoria, BC V8W 2Y2, Canada, hbliss@uvic.ca), and Bryan Gick (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

This paper reports on an initiative to develop a pronunciation training tool for learners of SENĆOŦEN (Salish: Canada). In acquiring new speech sounds, second language learners rely on both acoustic and articulatory information. Ultrasound has been demonstrated to be effective for pronunciation instruction, as it allows learners to visualize tongue movements during speech. However, ultrasound-based instruction can be difficult to implement with large groups or with learners wanting to learn independently. To address these limitations, we developed a technique for creating ultrasound overlay videos which combine ultrasound images of tongue movements with external profile views of a speaker's head. Ultrasound is particularly useful in SENĆOŦEN, as its inventory includes many lingual articulations that are difficult to distinguish without visual cues, such as velar/uvular contrasts, and a complex coronal series (e.g., tʼ, tʼ$^{\theta}$, ʧ, ʃ, ƛ̓, ɬ, l, lʼ). SENĆOŦEN is critically endangered, but efforts are underway to reawaken the language. Children participate in immersion programs, but parents need language-learning opportunities that are adaptable to their schedules and needs. We developed a series of videos for distribution to community members, in particular parents, giving them opportunities to improve their pronunciation to better communicate with their children in their heritage language.

4pSC64. Influence of age-related factors on English /r/-/l/ audiovisual training for Japanese speakers. Yasuaki Shinohara (Faculty of Sci. and Eng., Waseda Univ., Bldg. 51, 5th Fl., Rm. 08, 3-4-1, Okubo, Shinjuku-ku, Tokyo 169-8555, Japan, y.shinohara@aoni.waseda.jp)

Audiovisual training has proven successful for improving Japanese adults' identification accuracy of the English /r/-/l/ contrast. However, its effects on Japanese children have not been investigated yet. In the present study, 10 sessions of audiovisual English /r/-/l/ identification training were given to Japanese adults and children, and the age effects on learning the contrast in three testing conditions (audio-only, visual-only and audiovisual) were examined. It was hypothesized that children would improve their visual perception more than adults and they would show more audiovisual advantage than adults (i.e., higher improvement in the audiovisual condition than in the audio-only and visual-only conditions). The results demonstrated that both adults and children improved their identification accuracy in all three conditions, and the increase of the identification accuracy in the audiovisual condition was higher than that in the audio-only condition. However, there was no significant difference in improvement between adults and children. Due to a possible ceiling effect, it was unclear whether children had an advantage over adults in integrating visual information to auditory perception.

4pSC65. Acoustic analysis on before- and after-training English speeches by a male Japanese high-school student. Kikuyo Ito (National Inst. of Technol., Kagawa College, 355 Chokushi-cho, Takamatsu, Kagawa 761-8058, Japan, kikuyoito@hotmail.com)

Two sets of a 7-minute English speech by a 15-year-old Japanese male speaker, one made before two-month intensive training (approx. 2 hours per day, 3 to 4 days per week) and the other made after the training, were recorded and acoustically analyzed. The speech manuscript was constructed for a national English speech competition for students of the National Institutes of Technology in Japan, with ages ranging from 15 to 22 years. The intelligibility ratings of the two speeches evaluated by native speakers of American English (AE) indicated that the post-training speech was much more intelligible than the pre-training speech. Preliminary acoustic analysis focusing on the most frequently occurring words "traffic accidents" indicated that the temporal alignments of the consonant-vowel sequences of the words, along with the intensity and the duration of the stressed and unstressed vowels, were evidently different between the two speeches, with the tokens in the post-training speech closer to those in the speech by a native AE speaker than those in the pre-training speech. To identify the acoustic properties that make non-native speech "better sounding," three-way comparisons between the pre-training speech, the post-training speech, and the native speech are discussed in detail.

4pSC66. Auditory and acoustic-phonetic mechanisms of adaptation in the perception of sibilant fricatives. Eleanor Chodroff and Colin Wilson (Cognit. Sci., Johns Hopkins Univ., Krieger Hall 237, 3400 N. Charles St., Baltimore, MD 21218, chodroff@cogsci.jhu.edu)

Previous research has demonstrated that speech perception is highly dependent on preceding acoustic context (e.g., Ladefoged & Broadbent, 1957), and suggested that this reflects adaptation to the long-term average spectrum (LTAS; e.g., Holt, 2006). The present study examined whether adaptation is further enhanced by acoustic-phonetic covariation among speech sounds within the natural class of sibilants. In each trial of Experiment 1, listeners were exposed to a [z]-initial CVC syllable and then categorized one member of an [s]-[ʃ] continuum. The distribution of the spectral center of gravity (COG) of [z] in the adaptation stimuli was manipulated (high vs. low). In Experiment 2, the [z]-initial syllables were replaced by LTAS-matched white noise. Listeners adjusted the [s]-[ʃ] boundary in accordance with the exposure conditions, showing contrastive adaptation in both experiments; however, the adaptation effect was significantly stronger with the speech precursor. Furthermore, in Experiment 1, fricative COG provided a better quantitative account of responses than COG of the full syllable (ΔBIC: 24). These findings indicate that while general auditory processes can give rise to adaptation, listeners also exploit acoustic-phonetic relations among sounds: here, high correlations among fricative COG means observed in a large multi-talker corpus ([s]-[z], $r = 0.62$, [s]-[ʃ], $r = 0.53$).

4p THU. PM

4pSC67. Adaptation and aftereffects in response to bite block perturbation. Brett Myers and Antje Mefferd (Dept. of Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 8310 Medical Ctr. East, Nashville, TN, brett.myers@vanderbilt.edu)

Although speakers can instantly adapt to the presence of a bite block (BB), they also refine their compensatory behavior over time. However, the extent to which BB perturbations elicit aftereffects and the mechanisms that contribute to refinement and aftereffects remain unknown. In this study, speakers belonging to either a practice or a no-practice group produced sentence repetitions under five conditions: PRE-BB (jaw-free), BB1 (jaw-fixed, initial BB exposure), BB2 (jaw-fixed, after 20 minutes of BB exposure), POST1 (jaw-free, immediately after BB removal), POST2 (jaw-free, one minute after BB removal). All speakers held a 10mm BB in place for 20 minutes with the practice group reading aloud and the no-practice group sitting quietly. Jaw and posterior tongue kinematics were examined during the vowel /i/ embedded in the word "please" using electromagnetic articulography. Preliminary findings (14 speakers) revealed a significant main effect of condition only. That is, relative to PRE-BB, tongue height did not significantly change during BB1 and BB2 (indicating successful adaptation), but significantly increased during POST1 and POST2 (indicating an aftereffect). Jaw height remained unchanged from PRE-BB to POST1 and POST2 (indicating no aftereffect). Findings will be discussed with regard to internal models and the roles of proprioceptive and acoustic feedback.

4pSC68. Generalization in VOT imitation: Feature adaptation or acoustic-phonetic covariation? Colin Wilson, Eleanor Chodroff (Dept. of Cognit. Sci., Johns Hopkins Univ., Baltimore, MD), and Kuniko Nielsen (Linguist Dept., Oakland Univ., 1025 HHB, Rochester, MI 48309-4401, nielsen@oakland.edu)

After hearing instances of a word initial voiceless stop with lengthened VOT (e.g., long-[p^h]), speakers lengthen their VOTs in unheard words beginning with the same stop ([p^h]) and, critically, in words beginning with a different stop ([k^h]; Nielsen, 2011). This pattern of generalized phonetic imitation has previously been attributed to talker adaptation at the level of distinctive features or gestural organization. Implicit in these accounts is the assumption that talkers implement the same feature or gesture in a uniform way: generalization from long-[p^h] to long-[k^h] is an effective and rational adaptation strategy only if VOT values for different stops *covary* across talkers. Building on previous findings (e.g., Theodore, 2009; Chodroff & Wilson, 2015), we establish that covariation is robust in the laboratory speech of Nielsen (2011). In baseline productions, the mean VOTs of [p^h] and [k^h] were almost perfectly correlated across participants ($r = 0.97$, $p < .001$). Following familiarization the correlation remained strong but was somewhat reduced ($r = 0.87$, $p < .001$), reflecting clear but imperfect generalization of VOT lengthening. These results are consistent with abstract adaptation accounts targeting features or gestures, but suggest a novel model of VOT adaptation that directly exploits the empirical correlations between stops.

4pSC69. Phonetic convergence in an immersive game-based task. Tifani M. Biro, Joseph Toscano (Psych. Dept., Villanova Univ., Tolentine Hall, 800 E Lancaster Ave., Villanova, PA 19085, tbiro@villanova.edu), and Navin Viswanathan (Speech-Language-Hearing: Sci. & Disord., Univ. of Kansas, Lawrence, KS)

Phonetic convergence occurs when talkers change the acoustic-phonetic characteristics of their speech to be more similar to a conversational partner. It is typically studied using laboratory tasks, but the extent to which talkers converge varies considerably across studies. One aspect of these tasks that differs from real-world settings is how engaging they are. Highly contrived tasks may fail to elicit natural speech production, which could influence whether or not talkers converge. We address this issue by comparing the extent to which interlocutors converge in a repetitive, unengaging task versus an immersive video game-based task. Both tasks were designed to elicit production of specific words. Thirty word-initial voicing minimal pairs were used as stimuli, and we measured the degree to which phonetic cues (e.g., voice onset time; VOT) changed over the course of the experiment. For the more engaging task, participants' VOT values for voiceless tokens trended towards convergence (i.e., they gradually shifted their VOTs to be more similar to their partner's speech). In contrast, we found no clear evidence for convergence in the less engaging task. These results suggest that engaging, naturalistic tasks may yield results that more accurately reflect real-world conversational speech and phonetic variation than traditional laboratory experiments.

4pSC70. Acoustic and articulatory measures of prosodic convergence in a cooperative maze navigation task. Yoonjeong Lee, Samantha Gordon Danner (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, yoonjeol@usc.edu), Benjamin Parrell (Linguist, Univ. of Delaware, Newark, DE), Sungbok Lee, Louis Goldstein, and Dani Byrd (Linguist, Univ. of Southern California, Los Angeles, CA)

Convergence effects have been found in many aspects of communication between speakers in conversation. This study examines how phonetic properties of each speaker's prosodic structure differ before, during and after participating in a cooperative maze task. Using two NDI electromagnetic articulatory tracking systems simultaneously, we recorded speakers of American English in dyads (one male dyad & one female dyad). Acoustic measurements included: 1) sentence durations, 2) phrase-final lengthening, 3) occurrence frequency of sentence-final boundary tones (percentage H%), and 4) peak F0 values in sentence-final words. Articulatory boundary strength measures for word-final consonant(s) articulations included: 1) movement durations, 2) displacement, and 3) time-to-peak velocity in phrase-medial and phrase-final positions. Speakers exhibited distinct phonetic patterns before the cooperative maze task. However, conversing speakers showed convergence effects in prosodic boundary strength, choice of boundary tone, and quantitative characteristics of boundary tone. Convergence effects persisted in the speech of one or both members of each dyad even after conversation has ended. Our findings raise the possibility that qualitative and quantitative aspects of prosody can be modeled as arising from a single system of control and that this system of prosodic structuring can show coupling effects across speakers. [Work supported by NIH.]

4pSC71. Acoustic correlates of vocal alignment to accented speech. Eva M. Lewandowski (Psych., Emory Univ., 36 Eagle Row, Dept. of Psych., Ste. 270, PAIS Bldg., Atlanta, GA 30306, elewand@emory.edu) and Lynne C. Nygaard (Psych., Emory Univ., Alanta, GA)

Language users can vocally align to the speech characteristics of other talkers both within and across accent groups. Although perceptual assessments of vocal alignment indicate convergence to accented speech, the acoustic properties of the speech signal that contribute to these perceptual judgments are less clear. The present investigation examined the acoustic correlates of vocal alignment to foreign-accented and native speech and related acoustic to perceptual measures of convergence. Native speakers of American English shadowed utterances produced by four model talkers; male and female native speakers of English and Spanish. Baseline utterances were obtained from visual naming blocks and shadowed utterances were compared to both baseline and model talker utterances. Fundamental frequency (F0), utterance duration, and vowel spectra (F1, F2) served as acoustic indices of vocal alignment. The results showed that different patterns of alignment emerged across the three measures. Utterance duration, but not fundamental frequency or vowel spectra, demonstrated convergence. However, both duration and vowel spectra measures significantly predicted perceptual assessments of vocal alignment. The results suggest that perceptual judgments of alignment behavior depend on multiple acoustic properties and that the dimensions of acoustic change during vocal alignment appear similar across accented and native speech.

4pSC72. Explicit imitation and working memory. Miranda R. Morris (Linguist, UC Davis, 1 Shields Ave., Davis, CA 95616, mirmor@ucdavis.edu)

Much research a has been conducted in regards to implicit imitation, that is, phonetic imitation that occurs largely without a speaker's knowledge. However, explicit imitation, which occurs intentionally, has received far less attention. While it is observed that individuals vary in explicit imitative abilities, i.e., some individuals' speech shift more towards a given target voice than others, the factors that contribute to these differences have yet to be identified. The current study investigates individual differences in explicit imitation by exploring the correlation between explicit imitation and working memory, the memory involved in short term conscious processing. Subjects

participated in a baseline word reading task as well as an imitation task which involved repeating the same words aloud after hearing prerecorded stimuli from a Texan speaker. Additionally, subjects completed a reading Span task in order to measure working memory capacity and determine the effect of working memory on degree of explicit phonetic imitation. The degree of proportional phonetic shift towards the target speaker was correlated with Span scores in order to determine what link exists between explicit imitation and working memory on the level of the individual.

4pSC73. An experimental investigation of acoustic task-based carryover effects. Rachael Tatman (Linguist, Univ. of Washington, Guggenheim Hall, 3940-2425 Benton Ln., Seattle, WA 98195, rctatman@uw.edu)

Carryover effects, where conditions in one experimental task affect behavior in subsequent tasks, have not received much attention in linguistics. This is worrying, given that linguistic elicitation tasks do affect talkers' production (Labov *et al.* 1972, Warner *et al.* 2012). This study used before and after within-subjects design. 16 talkers completed three tasks: a memory task, one of a set of intermediate tasks (passage reading, word reading or an interview), and a second memory task.Pitch, intensity and duration of stressed vowels were measured for productions from each task. Nested mixed linear effects models were constructed for each measure. In all models, acoustic measures were used as the response variable and subject and token as random intercepts. For data from the intermediate tasks, the inclusion of task identity significantly improved model fit for pitch ($p < 0.05$), intensity ($p < 0.001$) and duration ($p < 0.001$). In other words, there were very strong task effects. However, when comparing data from the first and second memory task, including the intermediate task did not improve model fit for pitch ($p = 0.71$), intensity ($p = 0.78$) or duration ($p = 0.70$); these strong task effects did not carry over.

THURSDAY AFTERNOON, 1 DECEMBER 2016 SOUTH PACIFIC 1, 1:30 P.M. TO 2:50 P.M.

Session 4pSPa

Signal Processing in Acoustics: High Resolution Imaging Sonars Including Real Aperture, Synthetic Aperture, and Tomographic Sonars II

Brian G. Ferguson, Cochair
DSTO, PO Box 44, Pyrmont, NSW 2009, Australia

Timothy Marston, Cochair
APL-UW, 1013 NE 40th Street, Seattle, WA 98105

Invited Papers

1:30

4pSPa1. Tomographic sonar imaging of underwater objects and the excitation of structural waves. Brian G. Ferguson (Maritime Div., DSTG, PO Box 44, Pyrmont, NSW 2009, Australia, Brian.Ferguson@dsto.defence.gov.au) and Ron J. Wyber (Midspar Systems, Oyster Bay, NSW, Australia)

Tomographic sonar imaging (or image reconstruction from projections) requires insonifying the object to be imaged with a wideband monostatic sonar over a complete set (360^0) of look angles (or aspect angles). The tomographic sonar image that is reconstructed from the projection data (variation with aspect angle of the sonar impulse response) represents the projection of the object's three-dimensional acoustic reflectivity function projected on the imaging plane. The tomographic reconstruction of the backscattered sonar signals reveals the geometrical shape of the object, together with any prominent features on the object like lifting lugs. However, the formation of the tomographic image can be adversely affected by the presence of structural waves that can be excited by the insonification of the object at certain aspect angles. It is shown that a variety of these structural waves are readily observed in the projection data space where they are resolved in both time and aspect angle. The mapping of these structural waves to particular featues in the tomographic sonar image is demonstrated. The presence of these structural waves provides another means of classifying the object that complements sonar imagery.

1:50

4pSPa2. Circular synthetic aperture sonar images and aspect dependent spectral properties of solid elastic cubes. Viktor Bollen, Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, viktor.bollen@wsu.edu), and Daniel S. Plotnick (Acoust., Appl. Res. Lab, Seattle, WA)

While cubes may not be commonly found underwater, studying their aspect and material-dependent backscattering mechanisms may be useful in leads to understanding the scattering physics of other objects. Two solid cubes, made of either steel and or brass, were insonified in water using a circular synthetic aperture sonar (CSAS) system; Fourier based techniques were used to reconstruct target images. Identifiable backscattering mechanisms include edge-diffraction and elastic responses. Rayleigh waves, a class of surface elastic waves, are excited near a material specific incident angle (Rayleigh angle) that depends on the material of the cube. Those waves may

retroreflect from the edges of the cube's facee edges of the cube's face in such a way as to enhance backscattering [K. Gipson and P. L. Marston, J. Acoust. Soc. Am. **105**, 700-710 (1998)]. The steel cube displayed a strong Rayleigh wave response, easily visible in CSAS images; for the smaller brass cube that said mechanism was relatively subtle. Aspect dependent spectral properties can bewere extracted from the images or directly from the time-domain records. When the cube's top ridge is slanted vertically, an aspect angle dependent splitting in the low frequency spectrum can bewas observed and associated with edge diffraction. The experimental results were compared to Kirchhoff-Integration based simulations, which did notthat don't include elastic responses. [Work supported by ONR.]

2:10

4pSPa3. Sonar image reconstruction of objects near a reflective boundary. Daniel Plotnick (Appl. Phys. Lab., Univ. of Washington, 1510 NW Turner Dr., Apt. 4, Pullman, WA 99163, dsplotnick@gmail.com), Philip L. Marston (Washington State Univ., Pullman, WA), and Timothy M. Marston (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The properties of reconstructed acoustic images, created from data obtained using backscattering synthetic aperture sonar (SAS) systems, have been studied for objects in the free field. The presence of a nearby reflecting boundary, such as the seafloor or surface, greatly increases the number of mechanisms or raypaths by which sound may be backscattered to the sonar system. Some of these raypaths involve one or more interactions with the reflecting boundary (multipaths), and others involve sound scattering multiple times from the same target (multiple scattering). Experiments were conducted using an air-water interface and simple cylindrical targets in order to examine the properties of the reconstructed circular SAS (CSAS) images, including effects due to vertical obliquity of the target [D. Plotnick *et al*., J. Acoust. Soc. Am. **137**, 470-480 (2015)]. The CSAS image is by definition a two-dimensional representation of three-dimensional scattering; the image is thus highly dependent on the object's 3-D shape, position, and orientation; many of these image properties may be understood. An additional experiment was conducted using a linear SAS system for two closely spaced cylinders in the free field, allowing the effects of multiple scattering to be examined. [Work supported by ONR.]

2:30

4pSPa4. Quality assessment of acoustic color signatures. Daniel A. Cook (Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, dan.cook@gtri.gatech.edu), J. D. Park (The Penn State Univ. Appl. Res. Lab., State College, PA), and Alan J. Hunter (Univ. of Bath, Bath, Somerset, United Kingdom)

Recent years have witnessed increasing research by the minehunting community into the use of wideband, widebeam, low frequency sonar, typically below 50 kHz, whose purpose is to excite and observe an object's structural response. These signatures exhibit uniqueness allowing mines to be more clearly distinguished from clutter objects. Considerable effort has been directed toward studying the scattering physics, but less work has been done to optimize the signal processing used for extracting signatures from data collected at sea under realistic conditions. Specifically, the tradeoff between signals, sources of noise, and signal processing parameters is not well understood. For example, a long observation interval can capture structural acoustic effects occurring later in time than the geometric scattering from an object, at the expense of increasing the interference from sea floor reverberation. Relationships such as this are described, and metrics are suggested for choosing the most appropriate data collection and signal processing strategies.

Session 4pSPb

Signal Processing in Acoustics: High Resolution Imaging Sonars Including Real Aperture, Synthetic Aperture, and Tomographic Sonars III

Brian G. Ferguson, Cochair
DSTO, PO Box 44, Pyrmont, NSW 2009, Australia

Timothy Marston, Cochair
APL-UW, 1013 NE 40th Street, Seattle, WA 98105

Invited Papers

3:35

4pSPb1. Three-dimensional image reconstruction of objects using synthetic aperture sonar. Daniel Plotnick (Appl. Phys. Lab., Univ. of Washington, 1510 NW Turner Dr., Apt. 4, Pullman, WA 99163, dsplotnick@gmail.com) and Timothy M. Marston (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Synthetic aperture sonar imaging is typically considered for the case of linear or circular scans, where the reconstructed acoustic image is a two-dimensional representation of the true three-dimensional scattering from some object. However, a multi-dimensional aperture may be used to create a 3-D or volume image reconstruction of the acoustic scattering by that object. The 3-D image reconstruction may be performed using a delay-and-sum algorithm, or using comparatively fast wavenumber based methods. Simulated point scatterers will be used to initially demonstrate the algorithms, followed by a demonstration involving a simple object near a reflecting boundary. For objects that are located close to a reflecting boundary there exists a rich 3-D scattering structure. When imaged in two dimensions, object features becomes distorted due to the projection of scattering loci (e.g. the broadside reflection from a cylindrical target) that may be located outside of the image plane [Plotnick *et al.*, J. Acoust. Soc. Am. **139**, 2053 (2016)]. Volume images of such targets move these scattering loci to new locations in 3-D space, which may aid in understanding the target's shape and orientation. The point spread function and 3-D sampling requirements will also be considered.

3:55

4pSPb2. Scan geometries for three dimensional synthetic aperture sonar tomography. Timothy Marston (APL-UW, 1013 NE 40th St., Seattle, WA 98105, marston@apl.washington.edu) and Jermaine L. Kennedy (NSWC-PCD, Panama City Beach, FL)

Synthetic aperture sonar (SAS) exploits vehicle motion and coherent multi-ping signal integration to enhance image resolution. SAS is typically used to generate high-resolution 2D sonar imagery; however the generation of high-resolution, voxel-based 3D images of targets has also been demonstrated by synthesizing multi-dimensional apertures from scans conducted at diverse grazing-angles. In the latter case, the geometry of the multi-dimensional synthetic aperture can have significant ramifications for all stages of the beamforming process, ranging from navigation refinement and beamformer design to very practical experimental issues, such as the time necessary to scan a target. In this paper, the effects that different multi-dimensional scans have on three-dimensional SAS are considered. Results of field tests for different scan geometries are shown and interpreted in light of their various benefits and detriments.

4:15

4pSPb3. The Advanced Synthetic Aperture Sonar Imaging eNgine (ASASIN), a time-domain backprojection beamformer using graphics processing units. Isaac Gerg (Signal Processing, Penn State Appl. Res. Lab, 120 Forest Glen Circle, Port Matilda, PA 16870, idg101@arl.psu.edu)

We present a synthetic aperture sonar beamforming engine called ASASIN. ASASIN is a graphics processing unit based time-domain backprojection beamformer for Windows and Linux utilizing the NVIDIA architecture. It supports arbitrary array geometries and has a modular data input system. ASASIN uses one or more graphics processing units to beamform synthetic aperture sonar data faster than real-time. Its interface is suitable for operators and research scientists. In this talk we discuss ASASINs algorithms and computational performance along with the choices confronted (both algorithmic and computational) in its design. Finally, we discuss its computational performance across a variety of NVIDIA graphics processing units including their embedded models.

4:35

4pSPb4. Moving away from the phase center approximation in micronavigation for synthetic aperture sonar. Isaac Gerg (Commun., Information, & Navigation Office, Penn State Appl. Res. Lab, 120 Forest Glen Circle, Port Matilda, PA 16870, idg101@arl.psu.edu)

In many respects, the difficult problem of synthetic aperture sonar is motion estimation of the platform since there exists no reasonably priced inertial measurement unit which can meet the location accurately requirements needed to generate high resolution imagery. Many beamforming codes estimate their motion using a displaced phase center technique. This technique is popular but makes some approximations that break down in long range systems and in significant motion environments. In this paper, we discuss the motion estimation algorithm used in ASASIN, a time domain back projection beamformer developed at Penn State Applied Research Laboratory. Our motion estimator assumes displaced phase centers but moves away from the phase center approximation. Rather than estimating the vehicle position delta ping-to-ping independently, we estimate the vehicle's velocity and acceleration through all pings in 3 space. We use Google's Ceres solver to solve the resulting non-linear equations and additionally add a regularization term to deal with missing and bad data samples. Finally, we show imagery formed using our algorithm.

Contributed Paper

4:55

4pSPb5. Noncontact measurement of body position and vital information using airborne ultrasound. Kotaro Hoshiba (Dept. of Systems and Control Eng., Tokyo Inst. of Technol., W8-30 2-12-1 Ookayama, Meguro-ku, Tokyo 152-8552, Japan, hoshiba@cyb.mei.titech.ac.jp), Kazuhiro Nakadai (Dept. of Systems and Control Eng., Tokyo Inst. of Technol., Honda Res. Inst. Japan Co., Ltd., Meguro-ku, Tokyo, Japan), Shinnosuke Hirata, and Hiroyuki Hachiya (Dept. of Systems and Control Eng., Tokyo Inst. of Technol., Meguro-ku, Tokyo, Japan)

Because of a growing need for unconstrained medical monitoring for unobtrusively observing individuals in a living space, we have been studied about non-contact measurement of vital information such as respiration and heartbeat using airborne ultrasound. In previous study, the measurement system of small displacement using the M-sequence-modulated signal and tracking phase difference of reflected signals from the target has been proposed. The measurement of respiration and heartbeat of the target in a standing position using a pair of the loudspeaker and the microphone has also been performed. However, body position cannot be measured using a pair of the loudspeaker and the microphone because the system can measure the distance to the body only. In this paper, we describe a basic study of the measurement of body position, respiration, and heartbeat. In the system, using microphone array, body position was estimated by synthetic aperture processing. In addition, small body-surface velocity by respiration and heartbeat were measured by tracking phase difference of reflected signals.

THURSDAY AFTERNOON, 1 DECEMBER 2016 — NAUTILUS, 1:00 P.M. TO 4:25 P.M.

Session 4pUWa

Underwater Acoustics, Signal Processing in Acoustics, and Acoustical Oceanography: Harbor Acoustics II

Peter J. Stein, Cochair
Scientific Solutions, Inc., 99 Perimeter Road, Nashua, NH 03063

David L. Bradley, Cochair
Penn State University, PO Box 30, State College, PA 16870

Chair's Introduction—1:00

Invited Paper

1:05

4pUWa1. High-frequency seafloor scattering in a dynamic harbor environment: Observations of change over time scales of seconds to seasons. Thomas C. Weber and Larry Ward (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, tom.weber@unh.edu)

Predicting sonar performance in a harbor environment can be made challenging by the dynamics of the upper and lower surface boundaries. In this talk we examine measurements of the seabed in Portsmouth Harbor, NH. Tidally influenced currents in the Piscataqua River make Portsmouth Harbor one of the most challenging commercial ports to navigate in the northeastern United States. The level of

interaction between these currents and the harbor floor is a function of the substrate type. Stereo-camera observations in a sand-wave field near the harbor entrance show fluctuations in microscale roughness and optical reflectance at time scales of seconds during periods of high current. Despite the observed microscale dynamics in the sand-wave field, acoustic observations of both high-frequency seabed scattering strength and mesoscale topography in the same area appear stationary over time scales up to seasons. The low-level of observed fluctuations in scattering strength from the sand-wave field are commensurate with the gravel (and presumably less mobile) river thalweg. Nearby bedrock and sand seafloors show similarly low fluctuations over large time scales, despite seasonal variations in benthic fauna.

Contributed Paper

1:25

4pUWa2. Laboratory measurements of a phase shift in the reflection from a water-mud interface due to water column variability. Gabriel R. Venegas and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712-1591, gvenegas@utexas.edu)

Harbor basins and certain continental shelf environments experience significant variability due to tides, surface heating, currents, and other oceanographic processes. Dynamic processes such as these can cause substantial sound speed and density fluctuations in the water column over short time periods, which are often unknown to sonar users. Such variations are not critical over high-velocity bottoms such as sand, but can produce significant changes in how sound interacts with low-velocity fine-grained sediments. High levels of variability and a low-velocity bottom, such as mud, can therefore present challenges in applications including mine detection, port protection and shallow water sonar. At certain penetration angles, temporal variations in the water-to-sediment sound speed ratio can cause 180 degree phase shifts in the reflected field, which in turn produces waveguide propagation and target scattering field variability. To begin to understand these processes, laboratory measurements of plane wave reflection were obtained from a water-mud interface while varying salinity and temperature of the water. Results indicate that the dynamic nature of sound speed ratio at the ocean bottom can cause significant effects in shallow water environments with muddy bottoms. [Work supported by ONR.]

Invited Papers

1:40

4pUWa3. Two-stage active sonar network track-before-detect processing in a high clutter harbor environment. Geoffrey S. Edelson (Maritime Systems & Technol., BAE Systems, MER15-2350, P.O. Box 868, Nashua, NH 03061-0868, geoffrey.s.edelson@baesystems.com)

Reliable active acoustic detection and tracking of small targets while minimizing the number of false alerts is challenging in a shallow, multipath-inducing, high-clutter harbor environment. These targets often exhibit low target strength and the acoustic clutter fields can be dense and highly dynamic. One approach to detect and track targets in this environment involves a two-stage tracker due to the processing gain required to continually track a weak target in such a significant clutter field. The first stage uses more than a single frame of observations to not only initiate tracks but also to contribute to track maintenance. The second tracking stage is initiated by the tentative target tracks extracted from the first stage. Track segment association algorithms are employed to combat periods or locations track intermittency that inhibit detection and cause tracks to fail and restart. This paper describes the results of this two-stage approach to track-before-detect processing applied to a networked harbor surveillance active acoustic detection and tracking system that consists of a large number of relatively simple nodes, employs a windowed Hough-Transform tracker at the beam level, and a Kalman-based, multi-object tracker at the single- and multi-node levels.

2:00

4pUWa4. Underwater threat detection and tracking using multiple sensors and advanced processing. Timothy W. Acker (BioSonics, Inc., 4027 Leary Way NW, Seattle, WA 98107, tacker@biosonicsinc.com) and Andy Meecham (Northern Defense Industries, Alexandria, VA)

The vulnerability of military installations and critical infrastructure sites from underwater threats is now well accepted and, in order to combat these security weaknesses, there has been growing interest in—and adoption of—sonar technology. The well known challenges of the underwater environment, particularly in a harbor or port setting, can lead to a high Nuisance Alarm Rate (NAR). This, in turn, can lead to a lack of confidence from end users and a possibility that "real" alerts are incorrectly dismissed. In the past, this has been addressed by increasing the capability of individual sensors, leading to ever-increasing sensor complexity, however, the relationship between sensor performance and complexity/cost is highly non-linear. Even with the most complex and capable sensors, the fundamental limit to performance is often limited by acoustics, not sensor capability. In this paper, we describe an alternative approach to reducing NAR and improving detection of difficult targets (e.g., UUV's), through intelligent combination and fusion of outputs from multiple sensors and data/signal processing algorithms. We describe the statistical basis for this approach, as well as techniques, methodologies and architectures for implementation. We describe the approach taken in our prototype algorithms/system, as well as quantitative and qualitative results from testing in a real-world environment. These results show a significant reduction in NAR and increase in classification/alert range.

2:20

4pUWa5. Use of integrated instrumentation to detect and classify targets in shallow water. Emma D. Cotter, Paul Murphy (Mech. Eng., Univ. of Washington, Mech. Eng., UW Mailbox 352600, Seattle, WA 98195-2600, ecotter@uw.edu), James Joslin (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Steven Brunton (Mech. Eng., Univ. of Washington, Seattle, WA), Andrew Stewart (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Brian L. Polagye (Mech. Eng., Univ. of Washington, Seattle, WA)

The detection and classification of targets in shallow water environments (e.g., depth $<$ 20 m) is of increasing interest, both in security applications and more general environmental studies. Here, we report the ability of a cabled integrated instrumentation system to detect and classify targets in a shallow (15 m deep), narrow (150 m wide) channel influenced by moderate tidal currents (peak currents $<$ 2 m/s). The instrumentation system incorporates a passive acoustic array, multibeam sonar, acoustic camera, stereo-optical camera, and acoustic Doppler current profiler. The system effectiveness is benchmarked using autonomous "cooperative" targets (e.g., drifters with known position and acoustic characteristics), divers, and opportunistic observations of fish and marine mammals. By fusing data streams from multiple sensors, detection and classification rates are improved, particularly for smaller targets. Approaches to data management (data rates exceed 80 MB/s), real-time classification, and de-noising of acoustic imagery are also discussed.

2:40–2:55 Break

2:55

4pUWa6. Harbor passive acoustic monitoring systems. Philip Abbot, Vince Premus, Charles Gedney, and Mark Helfrick (OASIS, Inc, 5 Militia Dr., Lexington, MA 02421, helfrick@oasislex.com)

The feasibility of detecting quiet moving targets (e.g., unmanned undersea vehicles) in harsh harbor environments by using passive acoustic bottom-mounted hydrophone arrays, combined with advanced signal processing detection systems tuned to the target's radiated noise signature characteristics is presented. The harsh shallow water environment is in Narragansett Bay, Rhode Island where UUV operations have been conducted. The sonar equation study considers the use of high resolution, multi-channel hydrophone arrays, measured source levels of representative UUVs (they are quiet), measured noise levels in the harbor (can exhibit high levels), and corresponding array gain. The resulting detection ranges for quiet targets are presented along with estimated false alarm rates. It is shown that passive sonar systems can be a viable component or modality for an underwater harbor defense monitoring system.

3:15

4pUWa7. Stevens acoustic research in the Hudson River. Alexander Sutin and Hady Salloum (Maritime Security Ctr., Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu)

Stevens Institute of Technology conducted numerous acoustic tests in the Hudson River. The main instrument was Stevens Passive Acoustic Detection System (SPADES) that consists of a simple array with 5 hydrophones and that was initially developed for diver detection. Later, we collected a large library of acoustic signatures of various vessels and developed numerous algorithms for detection, tracking and classification. We present a brief review investigating acoustic applications in the Hudson River, including: Acoustic tomography tests were conducted for measuring the river currents. The difference of time of flight to emitter and receiver placed near the opposite bank of the River was used for calculating the current. Numerous tests were conducted for estimating the Impulse Response and its variation that can be used for prediction of reliability and operating distances for underwater acoustic communication. Noise measurements from passing boats were used for estimating the acoustic attenuation in a wide frequency band. Several other possible future researches may include: Time Reversal Acoustic Focusing for diver deterrence. Active/Passive acoustic fence for diver detection. Bubble curtain for the suppression of underwater noise from a pile drive.

3:35

4pUWa8. Passive and active sonar signal processing methods for port infrastructure protection and harbor security. Brian G. Ferguson and Kam W. Lo (Maritime Div., DSTG, PO Box 44, Pyrmont, NSW 2009, Australia, Brian.Ferguson@dsto.defence.gov.au)

High frequency passive and active sonar signal processing methods can be used in cluttered shallow water complex acoustic harbor environments for the detection, classification, localization and tracking of asymmetric threats including fast inshore attack craft, divers, underwater vehicles, and sea mines. For fast inshore attack craft, the event duration is short (tens of seconds) which necessitates the automation of the functions of a high frequency high resolution active sonar. The sonar also cues effectors that enable a layered defense capability to counter the threat. For wide area passive undersea surveillance, single hydrophones can be used for multipath passive ranging, a pair of hydrophones for direction finding, and an array of hydrophones for estimation of a complete set of target motion parameters. Also, it will be shown that the measurement of the time delays of the direct path and multipath arrivals at a single pair of hydrophones enables estimation of a complete set of target motion parameters for a surface vessel.

Contributed Papers

3:55

4pUWa9. Deep learning approach to passive monitoring of the underwater acoustic environment. Eric L. Ferguson, Rishi Ramakrishnan, Stefan B. Williams (Australian Ctr. for Field Robotics, Univ. of Sydney, Rose St. Bldg., Sydney, NSW 2006, Australia, e.ferguson@acfr.usyd.edu.au), and Craig T. Jin (Computing and Audio Res. Lab., Univ. of Sydney, Sydney, NSW, Australia)

A cost effective approach to remote monitoring of protected areas such as marine reserves, harbors and restricted naval waters, is to use passive sonar to detect, classify, localize, and track marine vessel activity (including small boats and autonomous underwater vehicles). This paper uses data from a single hydrophone mounted above the sea floor to compare a conventional cepstral analysis method with a deep learning approach for monitoring marine vessel activity. Cepstral analysis of the underwater acoustic data enables the time delay between the direct path arrival and the first multipath to be measured, which in turn enables estimation of the instantaneous range of the source (a small boat). However, this conventional method is limited to ranges where the Lloyd's mirror effect (interference pattern formed between the direct and first multipath arrivals) is discernible. It is shown that a Convolutional Neural Network (CNN) operating on cepstrum data with a regression output improves the single sensor ranging performance by considering additional multipath arrivals. To demonstrate the effectiveness of the approach, a surface vessel is ranged using the CNN and the results compared with the conventional method.

4:10

4pUWa10. Multi-agent acoustic noise field characterization using optimal control. Caitlin Bogdan (Boston Univ., 110 Cummington Mall, Boston, MA 02215, cbogdan@bu.edu), Andrew Wixom (Structural Acoust., Penn State Univ., Appl. Res. Lab., State College, PA), and J. G. McDaniel (Boston Univ., Boston, MA)

Investigations of acoustic-driven control schemes have stayed away from traditional Optimal Control methods because of the computational load and lack of direct solution methods. The method presented here overcomes these challenges and learns a spatially varying noise field by applying the Pontryagin Maximum Principle (PMP) to solve for optimal trajectories over a changing estimate of the noise field. The resulting control drives the agents to the regions of the estimated field with the most uncertainty. This optimization is conducted with multiple agents interested in the noise field for a single frequency as well as for multiple agents simultaneously optimizing over multiple frequencies of interest. The challenges of finding a solution to the mixed boundary value problems is achieved through an eigenvalue decomposition method. The selected cost function is designed to minimize time and thus bound error introduced by the open loop control derived through the PMP methodology. Simulations and preliminary experimental results are presented to demonstrate the effectiveness of this method. The resulting information from utilizing this method are useful in optimized sensor location problems, which are especially relevant for the large dynamic array system used to do the initial sensing. [Research supported by NAVSEA and Raytheon Company.]

THURSDAY AFTERNOON, 1 DECEMBER 2016 — NAUTILUS 1, 4:40 P.M. TO 5:40 P.M.

Session 4pUWb

Underwater Acoustics: Ocean Ambient Noise: Sources, Variation, and Characterization I

D. Benjamin Reeder, Chair

Oceanography, Naval Postgraduate School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943

Contributed Papers

4:40

4pUWb1. Spatial variation of ocean ambient noise spectrum in offshore China. Guoli Song (Key Lab. of Underwater Acoust. Environment, Inst. of Acoust.,Chinese Acad. of Sci., North Fourth Ring Rd., Haidian District, No. 21, Beijing 100190, China, songguoli13@mails.ucas.ac.cn), Xinyi Guo, Li Ma, and Qianchu Zhang (Key Lab. of Underwater Acoust. Environment, Inst. of Acoust.,Chinese Acad. of Sci., Beijing, China)

Ocean ambient noise always presents spatial difference because of the diversity of the noise source and the propagation condition. This paper analyzes the noise spectrum level (NL) of two stations measured in offshore China, and finds that the difference of the two NL is more than 10 dB with different variation for per octave. Through analyze the NL and the transmission loss level (TL) from the noise source to the measuring station, a correlation is found between the noise level difference (NLD) and transmission loss difference (TLD), with a correlation coefficient of 0.78 at the band of 50 Hz ~500 Hz. This is a strong proof to connect the spatial variation of NL with the varied propagation channel, and also suggests that at this band the ambient noise is closely related to the ship noise sources. In addition, the difference of the ship noise source level coincides with the number of vessels before and after the fishing moratorium period in China, which further explains the difference between the NL of two experimental stations are highly related to ships.

4:55

4pUWb2. Spatial dependence of very low frequency wind-generated noise in the sound channel. Stephen M. Nichols and David L. Bradley (Appl. Res. Lab., The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, smn5198@psu.edu)

Storm-driven non-linear interaction of ocean waves excites Very Low Frequency (VLF) (<5 Hz) sounds locally. As the surface wind speed increases, the slope of the VLF acoustic spectrum approaches approximately -20 dB/octave, corresponding to a similar behavior observed in the surface wave spectrum. The amplitude of the measured acoustic spectrum in this frequency region is strongly correlated with local surface wind speeds [J.

4p THU. PM

Acoust. Soc. Am. **139**(3), 1110 (2016)]. Due to the sound propagation conditions in the open ocean, it is expected that sound from distant storms will also provide a significant contribution to the measured sound field. The impact of distant meteorological activity on the local acoustic spectrum is examined by comparing the ambient noise recorded by the Comprehensive Nuclear-Test Ban Treaty Organization's (CTBTO) hydro-acoustic monitoring system to global records of surface wind speed, remotely sensed via satellite.

5:10

4pUWb3. Characterization of very low frequency ambient noise by principal component analysis. Stephen M. Nichols and David L. Bradley (Appl. Res. Lab., The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, smn5198@psu.edu)

Long-term acoustic monitoring datasets compose a dynamic mixture of many source types, creating a unique noise field in each measurement location and for each recording time. Principal Component Analysis (PCA), and the correlation matrix from which the Principal Components are derived, offer an effective means of estimating the fundamental acoustic contributors, and their portion of the noise field. An improved characterization of ambient noise data sets is discussed, by representing the sound field in a smaller parameter space of principal components, each component reflecting the spectral characteristics of a specific sound source (i.e. fin whale vocalizations). This procedure leads to a definitive parsing of the contributing source types in the measured spectra. The datasets used are recordings from the Comprehensive Nuclear-Test Ban Treaty Organization's (CTBTO) hydro-acoustic monitoring system.

5:25

4pUWb4. Comparison of scattering mechanisms for excitation of axial modes by surface sources. Mehdi Farrokhrooz and Kathleen E. Wage (George Mason Univ., 4450 Rivanna River Way PMB3740, Fairfax, VA 22030, mfarrokh@masonlive.gmu.edu)

The vertical spectrum of low-frequency ambient noise in deep water contains significant contributions from distant shipping. Distant noise is concentrated at angles around broadside that are associated with the low order modes. Near-surface sources cannot excite the low modes directly. Dashen and Munk suggest three mechanisms for transferring energy from high modes to low modes: internal wave (IW) scattering, downslope conversion, and excitation at high latitudes where the sound channel intersects the surface. Dashen and Munk conclude that slope conversion is the most likely due to the weakness of IW scattering and the lack of shipping traffic at high latitudes. Commenting on their IW analysis, Dashen and Munk note that they are not totally convinced by their theoretical results and admit that IW scattering may have a significant effect. This talk compares the IW scattering and downslope conversion mechanisms for transferring surface source energy to the low modes. IW scattering is modeled using transport theory [Colosi and Morozov, JASA 2009] and compared to parabolic equation and coupled mode simulations. Results show that IW scattering from a large number of ships can produce vertical spectra comparable to those obtained through downslope conversion. [Work supported by ONR.]

Plenary Session and Awards Ceremony

Michael R. Stinson, Chair
President, Acoustical Society of America

Kentaro Nakamura
President, Acoustical Society of Japan

Presentation of Certificates to New Fellows

G. Clifford Carter – For contributions to time delay estimation in the presence of source or receiver motion

Francesco L. di Scalea – For contributions to the theory and applications of ultrasonic guided waves

Truls T. Gjestland – For contributions to research and standards development on transportation noise effects on communities

Nathan J. McDannold – For contributions to therapeutic ultrasound

Robert J. McGough – For contributions to numerical modeling in medical ultrasound

Jennifer L. Miksis-Olds – For contributions to underwater acoustic noise research and the integration of acoustics into marine ecology

Joseph A. Sisneros – For contributions to the understanding of fish hearing

James A. TenCate – For contributions to nonlinear acoustics of earth materials

Blake S. Wilson – For the development and enhancement of cochlear implants

Introduction of Award Recipients and Presentation of Awards

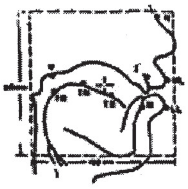
Medwin Prize in Acoustical Oceanography to Thomas C. Weber

Rossing Prize in Acoustics Education to Brad H. Story

Trent-Crede Medal to Earl G. Williams

Recognition of ASJ Organizers

Recognition of ASA Organizers



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PART ONE - RESPIRATION, PHONATION AND AERODYNAMICS

1. The whole body plethysmograph in speech research. John J. Ohala
2. Aerodynamic end respiratory kinematic measures during speech. Elaine T. Stathopoulos
3. Physiologically based models of phonation. Ingo R. Titze
4. Use of the electroglottograph in the laboratory and clinic. James J. Mahshie
5. Endoscopy, stroboscopy, and transillumination in speech research. Anders Lofqvist, Kiyoshi Oshima

PART TWO - INDIRECT ARTICULATORY MEASUREMENTS

6. Magnetic resonance imaging (MRI) in speech research. Carol Gracco, Mark Tiede
7. Imaging the tongue with ultrasound. Maureen Stone
8. Estimating articulatory movement from acoustic data. Kenneth N. Stevens
9. Electromyography in speech research, Kiyoshi Oshima. Katherine S. Harris, Fredericka Bell-Berti

PART THREE - DIRECT ARTICULATORY MEASUREMENTS

10. The rise and fall of the soft palate: The Velotrace. Fredericka Bell-Berti, Rena A. Krakow, Dorothy Ross, Satoshi Horiguchi
11. Dynamic electropalatography. William J. Hardcastle, Fiona Gibbon
12. Measuring articulatory movements with an electromagnetic midsagittal articulometer (EMMA) system. Joseph S. Perkell, Mario A. Svirsky, Melanie L. Matthies, Joyce Manzella
13. Optoelectronic measurement of orofacial motions during speech production. Eric Vatikiotis-Bateton, Kevin Munhall, David Ostry

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ACOUSTICAL SOCIETY OF AMERICA

TRENT-CREDE MEDAL

Earl G. Williams

2016

The Trent-Crede Medal is presented to an individual, irrespective of nationality, age, or society affiliation, who has made an outstanding contribution to the science of mechanical vibration and shock, as evidenced by publication of research results in professional journals or by other accomplishments in the field.

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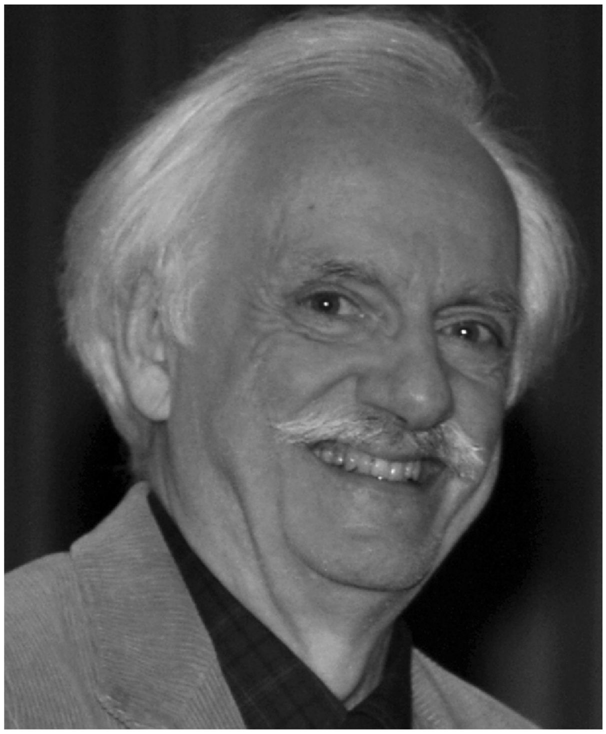

CITATION FOR EARL G. WILLIAMS

". . . for development and application of near-field acoustical holography"

HONOLULU, HAWAII • 1 DECEMBER 2016

Earl Williams' interest in acoustics is not incidental; he is an accomplished cellist, and his curiosity as to how the cello delivers the best sound to an audience led to a true passion for understanding the radiation of sound from a vibration structure. Earl received a Bachelor's degree in Electrical Engineering from the University of Pennsylvania in 1967 and a Master's degree from Harvard University in 1968, studying acoustics under the supervision of Ted Hunt. Earl continued his studies in acoustics with Eugen Skudrzyk at The Pennsylvania State University where he was awarded a Ph.D. in Acoustics in 1979. The title of his thesis was "Vibrations of Plates of Various Geometries."

After receiving his Ph.D., Earl remained at Penn State as a postdoc for J. D. Maynard. This collaboration led to the development of an array of microphones for measuring an acoustic "hologram", and data-processing algorithms, which demonstrated the remarkable power of nearfield acoustic holography (NAH) for measuring the sound radiation with an image resolution previously assumed to be impossible. The results of this research were published in *Physical Review Letters* (Vol. 45:554-57, 1980) and, with a more detailed description, in the *Journal of the Acoustical Society of America* (JASA) (Vol. 78:1395-1413, 1978); this latter paper is perhaps the most cited paper in the now extensive field of nearfield sound measurement.

At Penn State there was considerable NAH activity, with measurements on test sources, plain plates, plates with structure, musical instruments including Earl's cello, etc. Earl demonstrated a remarkable and rare ability: when confronted with a particular problem, he could identify the real crux of the problem, design and conduct appropriate experimental measurements and, having targeted the most important aspect, arrive at a comprehensive and valid understanding of the problem. This ability should be evident in the discussion of Earl's research accomplishments below.

With his Postdoctoral Scholar position having a finite duration, Earl sought a position elsewhere and received a number of job offers. Earl decided that a position with the Naval Research Laboratory (NRL) in Washington, DC, would provide the best opportunity for pursuing research; however, he probably stipulated that he would not be making sound radiation measurements on his priceless cello underwater. Earl began research at NRL in 1982, and it was there that he made some of the most important developments in sound radiation measurement.

The remarkable power of NAH arises because measurements in the nearfield of a source permit the detection of sound from modes which very weakly radiate from the source; such modes typically reveal the details of how sound is ultimately propagated from the source. A fundamental problem is that some modes are so weak that even in the nearfield their signals fall into the noise of the measurement system. This problem has two aspects: 1) identifying the weakly radiating modes, and 2) reducing the effects of the noise in the reconstruction process. In 1989, W.A. Veronesi showed that the first problem could be solved with singular value decomposition; this was followed by a flurry of activity trying to solve the second problem. Ideally one would want a standard, one-size-fits-all solution, but unfortunately there was an overabundance of solutions. Earl addressed this issue by undertaking a thorough, exhaustive survey of all the solutions and in 2001 published a paper in JASA (Vol. 110:1976-88), which could be used to identify the appropriate solution; this is perhaps the second most cited paper in the field.

Typically sound radiation studies are performed for structures which radiate outward, and one may assume the boundary condition at infinity of outward propagation only. However, there is also the important problem of sound radiation into the interior of a vibrating structure, such as the cabin of an automobile or aircraft. This is very difficult problem, but

Earl showed that useful measurements were possible, and published landmark results in JASA (Vol. 108:1451-63) in 2000.

A fundamental problem with measuring sources which radiate outward is that, theoretically, the measurement surface should enclose the entire vibrating structure in order to capture all of the radiated sound. Needless to say, this is a substantial problem for practical field measurements. Earl discovered that this theoretical restriction could be relaxed far more than one might expect and found that the measurement surface could in fact be considerably smaller than the size of the vibrating surface. Earl's crucial insight was that when measuring close to a vibrating (source) surface, the signal received near the center of the measurement surface would be dominated by the sources directly below. For the radiated field beyond the perimeter of the measurement surface one could make an educated approximation, and the reconstruction of the source surface just below the center of the measurement surface would probably be quite accurate. One could then use this reconstructed source to propagate out to get a new estimate of the field beyond the perimeter of the measurement surface, and this process could be repeated iteratively. This method, referred to as "patch" holography, is one of the most important practical extensions of nearfield measurement.

In 1999, Earl published a book, *Fourier Acoustics: Sound Radiation and Nearfield Acoustic Holography*, which made the theory and practice of NAH accessible throughout the acoustics community. Earl has made many more innovations other than the few discussed above, and his achievements have been recognized throughout the world as well as within NRL. No one has done more to make NAH a "household acronym" in the sound radiation and noise communities.

Earl has been recognized for his achievements by many organizations. He is a Fellow of the Acoustical Society of America (ASA) and an Associate Editor of JASA in Structural Acoustics. He has presented over 70 invited lectures at international meetings, many of them keynote lectures, and has presented 26 invited lectures and organized five special sessions at meetings of the ASA. His NAH research was formally recognized by NRL as one of the Navy's most innovative technologies over a period of 75 years. In 2002, Earl was awarded the distinguished rank of Senior Technologist at NRL. In 2003 he was invited to present the Rayleigh Lecture at a meeting of the American Society of Mechanical Engineers (ASME), and in 2009 was awarded the Per Bruel Gold Medal for Noise Control and Acoustics by the ASME.

It is most appropriate that Earl Williams be honored with the Trent-Crede Medal of the Acoustical Society of America.

J. D. MAYNARD
DAVID FEIT